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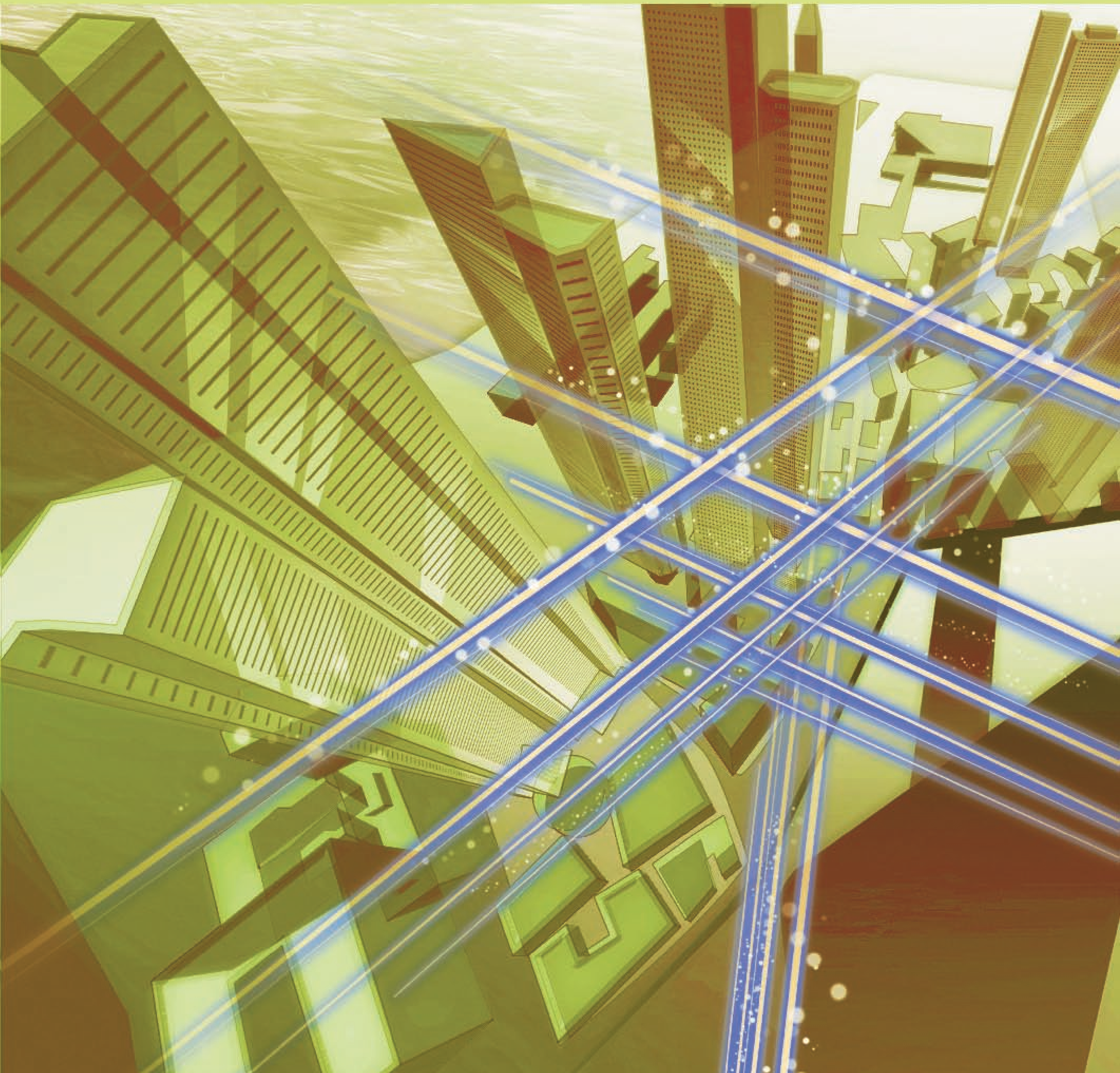

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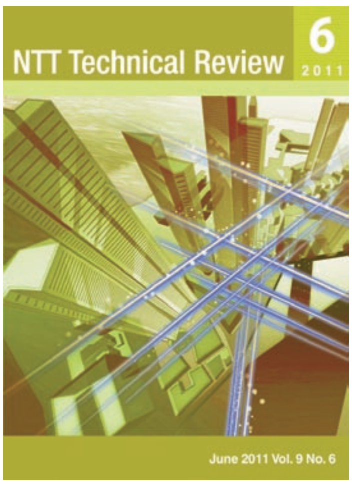

Feature Articles: Services for Hikari Era: Terminal Component Technologies

Regular Articles

Global Standardization Activities

Practical Field Information about Telecommunication Technologies

External Awards/Papers Published in Technical Journals and Conference Proceedings

Service and Terminal Component Technology Development Initiatives for the Optical Fiber Broadband Era (Hikari Era)

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Abstract

In the optical fiber broadband era (hikari era), which has just begun, optical fiber is being used to transmit large amounts of digital information at high speed. It enables the provision of a great variety of services to users and this variety will increase even more in the future. We introduce various research and development activities for terminal component technologies.

1. Introduction

The functionalities of terminals can be divided into three broad types: human interface functions, processing functions, and network functions. Our main aim for terminal research and development (R&D) in the optical fiber broadband era (hikari* era), which has just begun, is to make use of communications media, freely handle multiple types of media, have various types of media processing, and exploit all of these fully for the provision of many services of various types. For example, while the conventional communications terminal, an analog phone, can transmit only 3.4-kHz sound, hikari-era terminals will support a diverse range of medium types such as photographs, video, and data at any time as well as a diverse range within each medium (e.g., wideband or multichannel sound, high-resolution images, and video). Furthermore, in addition to supporting person-to-person communications, terminals will also need to exchange various types of data with servers. Hikari-era terminals will provide various services using large amounts of various media over the network for person-to-person and person-to-server communication.

2. Terminal component elements

Terminal functions can be divided into human interface, processing, and network functions, but these are implemented in either hardware or software. Terminal component elements are shown in **Fig. 1**.

Interface functions are implemented by a combination of the terminal shape and mechanisms that terminals are equipped with, touch panels, displays, handsets, keys, and other input/output devices.

Network functions can be implemented using software or with a network processor to utilize the high speed of hardware. In particular, in the hikari era, it must be possible to process high-volume media such as video (consisting of streams of large packets) without excessive delay and fixed-length packets such as VoIP (voice over Internet protocol) packets (within a delay of a few milliseconds, without any packet loss).

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* Hikari is the Japanese word for light.

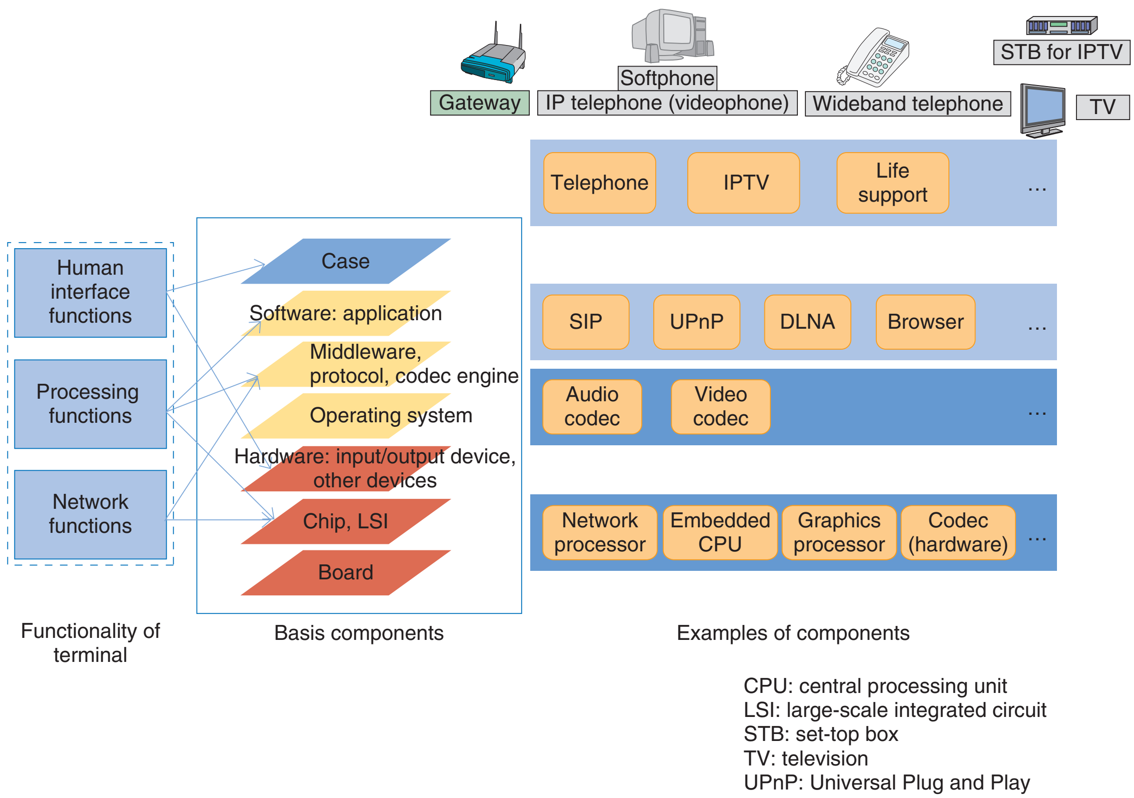

Fig. 1. Terminal component elements.

Because of this, network processor technology able to process packets at high speed is very important. The Feature Article in this issue entitled "Network Processing Technology for Terminals Enabling High-quality Services" [1] provides an introduction to these technologies.

Each processing function in a terminal is implemented with middleware and application software, but for sound, video, and other media processing in particular, the codec (coder/decoder) technology is very important. To implement a VoIP service with wideband sound, one must use appropriate audio codecs and input/output devices such as microphones and speakers to transmit and receive sound. Technology components for new hikari-era VoIP terminals that surpass ordinary 3.4-kHz analog sound are introduced in the article "Voice Processing Technology for Realistic-quality Voice and Rich Telecommunication Services" [2].

Moreover, in addition to audio codecs, video codecs are also important for communication by means of video. There are many possible applications for video codecs, from realtime bidirectional communication such as video-telephony to services distributing video from a server. Comfortable communication by video telephony and other bidirectional applications requires a low delay and high video quality, while services that distribute a variety of video content, such as IPTV (Internet protocol television) should have high video quality and a low distribution cost. The characteristics of the codec used must match the application. Video codec technologies for distributing video are described in the article "Software H.264 Encoder Engine for Online Video Delivery Service Cost Reduction" [3].

To enable the handling of various types of media, codecs alone are not enough. Also important are middleware, such as session initiation protocol (SIP) for call control functions and software based on Digital Living Network Alliance (DLNA) protocols for handling media, and specifications for how the codecs and middleware will be used when terminals communicate with each other. Specifications governing communications between terminals are called teleservice specifications. It is necessary to define the usage of low-layer network attributes as well as

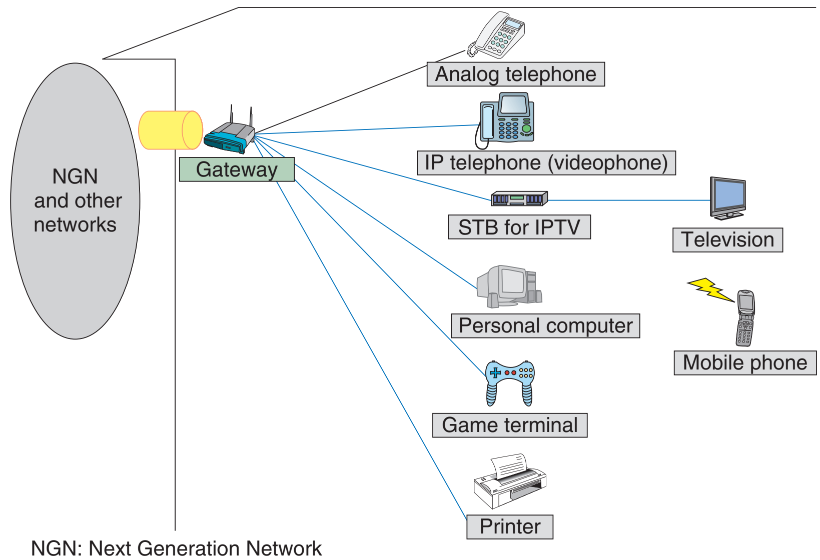

Fig. 2. Example of terminal connections in a home environment.

high-layer attributes available between terminals equipped with these network functions when they connect to each other. With ISDN (integrated services digital network), the main teleservices included telephony, fax, videotelephony, and telex, but in the hikari era, services will freely allow switching between various types of media in addition to the conventional ISDN teleservices. The article “Visual Softphone: New Ways to Communicate” [4] introduces the visual softphone as a terminal for the hikari era. The visual softphone can communicate using various types of data including photographs and video, while also handling voice and videophone communication.

The technical development of these various terminal components, from hardware to middleware, is becoming more important for achieving rich new services.

3. Home environments

In the hikari era, how will the various terminals be placed in the home and how will they be connected together? In the analog and ISDN eras, homes were wired with telephone and ISDN cables to connect telephones, but in the hikari era, optical fiber cable will be brought right to the home and connected to a home gateway, and various types of terminals will be connected to this home gateway. An example of terminal connections in a home environment is shown in **Fig. 2**. Multiple terminals will be connected and multiple types of media (video, sound, data, etc.) will be used over IP channels in addition to voice. Various existing IP communications technologies can also be utilized, from existing Ethernet cables to various wireless technologies (IEEE 802.11a, b, g, n, etc.), power-line communications, coaxial communications, etc.

With the increase in the number of choices for users and the ability to wire a home according to conditions there, it is becoming increasingly difficult to maintain the home environment. Wiring for conventional analog telephones or ISDN was simple, but in the hikari era, each room in a house can hold various types of terminal and each can be connected in various ways, creating a multitude of possibilities. Conventional office local area networks could be managed using protocols such as simple network management protocol (SNMP), but there is no standard management protocol like SNMP for home devices. This makes it difficult to determine what devices are connected to the network and the state of each device. Even though protocols such as Universal Plug and Play (UPnP) and DLNA provide some information about devices on the network, not all devices support these protocols, and the information they provide is limited. Therefore, we are developing a home network diagnosis tool [5] that makes it easier to obtain information about what devices are present and whether or not they are operating normally at the IP level by analyzing packets flowing between them on the network. This tool will enable engineers to detect problems in home networks and repair them quickly. Expert knowledge is generally required for network maintenance, and it is difficult to find highly skilled maintenance workers. This technology makes maintenance much easier by providing support functions such as rule-based interactive diagnostics and an easy-to-understand network map of connected devices designed for maintenance people with less experience.

4. Concluding remarks

We have provided a simple introduction to the state of R&D activities for terminal component technologies for the hikari era and home environments where these various terminals will be used. In the future, we will continue to conduct R&D of various services and terminal component technologies to promote the spread of these terminals in the hikari era.

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Voice Processing Technology for Realistic-quality Voice and Rich Telecommunication Services

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Abstract

This article describes the speech coding, acoustic design, and signal processing technologies for the high-quality telephone service that NTT provides over the Next Generation Network (NGN). Since the conventional telephone system uses a limited frequency range less than that of the human voice, voice individuality and clarity are sometimes insufficient. On the other hand, the high-quality NGN telephone service enables realistic-quality voice communication. A stereo conference phone under development for future services is also described.

1. Realistic-quality voice

In 2008, NTT launched a new IP (Internet protocol) telephone service called HIKARI DENWA*1 [1] over FLET'S HIKARI NEXT [2], its optical fiber subscriber line service. This is a telephone service based on voice-over-IP (VoIP) technology, and it optionally provides multimedia communications that enables a user to make calls in the same way as normal telephone calls. One of the optional services is a high-quality telephone service. Conventional telephones use only the human voice components in the frequency range from 300 Hz to 3.4 kHz. This limitation sometimes leads to insufficient individuality and similar words may be misheard. High-quality telephones use components in the range from 50 Hz to 7 kHz: twice the width of conventional telephones. This enables telecommunication with realistic-quality voice.

2. Speech coding technology

To enable speech signals to be transmitted over digital or packet networks, speech coding technology is used. The encoder and decoder together are called a codec. The legacy analog telephone system uses ITU-T G.711 codecs at subscriber switchboards (ITU-T: International Telecommunication Union, Telecommunication Standardization Sector). By contrast, in the HIKARI DENWA system, the codec is implemented in each terminal, i.e., an IP phone terminal or home gateway. When a user makes a normal telephone call over HIKARI DENWA, the G.711 codec is used, as in the conventional telephone service; when a user makes a high-quality telephone call, a wideband codec is used. The codec to be used for a telephone call is determined automatically by negotiation between the calling and receiving terminals by means of session initiation protocol (SIP) signaling. One well-known wideband codec is ITU-T G.722, which has been used for video and audio conferencing systems over ISDN (integrated services digital network) and IP networks and has recently begun to be used for IP phones.

Even when a wideband codec is used for a telephone call, the same functions as in a conventional

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*1 Hikari is the Japanese word for light; denwa is the Japanese word for telephone.

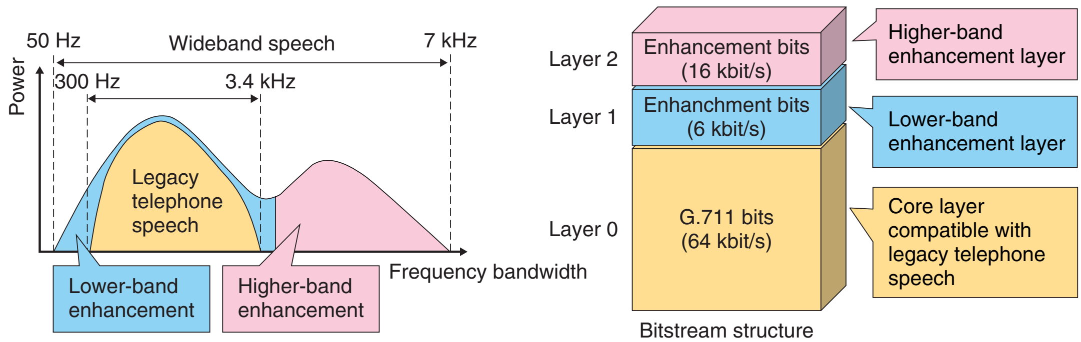

Fig. 1. Scalable coding.

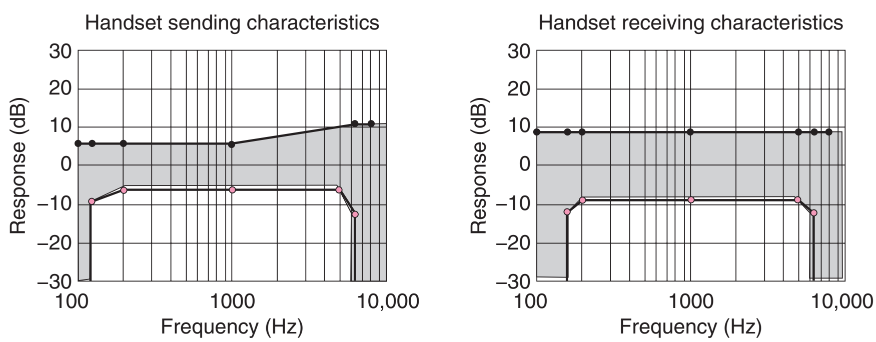

Fig. 2. Transmission characteristics for wideband telephone in CIAJ standards (CES-Q004).

telephone service, e.g., call waiting and call transfer, should be provided. However, codec negotiation is often problematic in this case. Codec negotiation was not considered before because there was only one codec: G.711. However, when G.711 and a wideband codec are used together, the codec should be switched appropriately.

To solve this problem, NTT Cyber Space Laboratories developed a scalable codec that extends the conventional G.711 and standardized it as ITU-T G.711.1. As shown in **Fig. 1**, this scalable codec has a multilayer structure consisting of a G.711 layer, lower-band enhancement layer, and higher-band enhancement layer. The G.711 layer is compatible with the conventional G.711 codec. This structure can be understood in terms of building blocks. G.711 and G.711.1 are easily converted to each other by extracting some blocks or by adding required blocks. Therefore, the use of G.711.1 enables wideband speech to be easily applied for conventional call functions, such as call waiting, call transfer, and multipoint teleconferencing [3].

3. Design strategy for high-quality handset

A handset is the most basic acoustic device for a telephone. A high-quality telephone cannot be made just by implementing wideband speech codec software. A normal handset is designed for narrowband (from 300 Hz to 3.4 kHz) speech, and it cannot pick up or reproduce over-3.4-kHz speech. We need a handset design for wideband speech and a standard for its characteristics.

The Communication and Information Network Association of Japan (CIAJ) standardized transmission characteristics for wideband (from 150 Hz to 7 kHz) digital handset telephones as CES-Q004 in 2007. CES-Q004 requires a wideband telephone to be able to send speech with a frequency range from 125 Hz to 6.3 kHz and receive speech ranging from 160 Hz to 6.3 kHz. The characteristics required for the handset are shown in **Fig. 2**.

We can use acoustic design knowledge for audio and broadcasting to design hands-free devices for communication using over-3.4-kHz speech. However,

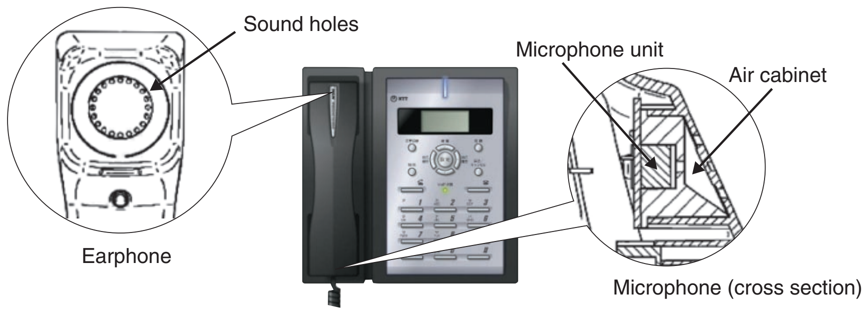

Fig. 3. High-quality telephone for NGN trial (NP-1).

this knowledge cannot be used for handsets: the design of a wideband handset is a new endeavor. The selection of a microphone and an earphone driver unit, the method of enclosing these units in a handset, noise suppression by circuit design, and equalization of frequency characteristics by signal processing are important for a high-quality handset design.

We developed the first high-quality telephone NP-1 for a trial of the Next Generation Network (NGN) in 2007. The handset uses microphone and earphone driver units intended for audio products rather than for telephones. The units were enclosed, and we made a small air cabinet in front of each unit. We adjusted the volume and shapes of the cabinets and the placement of sound holes for the earphone (**Fig. 3**). As a result, the handset achieved flat frequency characteristics up to 7 kHz. Generally, a flat-frequency earphone is more difficult to design than a flat-frequency microphone. The best design method differs according to the acoustic driver units that can be used. Therefore, when designing a handset, one must leave room for making sound holes and an air cabinet.

Noise suppression is not considered for the under-300-Hz or over-3.4-kHz frequency ranges in the circuits of a normal telephone. For wideband use, the circuits must be redesigned. If standard characteristics are not satisfied by the microphone and earphone designs, then the use of additional signal processing is an efficient measure.

We developed a high-quality Wi-Fi telephone as a prototype in 2009 (**Fig. 4**). It has a very small body for portability and does not have air cabinets. For a high-quality handset design, only large sound holes were added. The standard characteristics were satisfied by selecting an acoustic driver unit and by equalizing sound through signal processing.

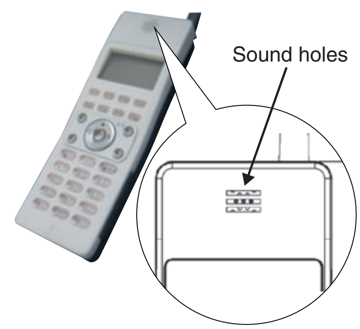

Fig. 4. High-quality Wi-Fi telephone.

4. Echo canceller

Hands-free speakerphones are useful for conversation and for sharing information among several people, while phone calls via a handset are suitable for one-to-one communications because the voice is heard only by the far-end speaker. Hands-free phones did not used to be widely utilized except for specific situations such as business teleconferences. In recent years, however, high-performance mobile terminals like netbooks or smartphones have become prevalent, and hands-free calls using those small devices have become more popular.

When you talk on a hands-free telephone (**Fig. 5**), the voice of the far-end speaker emitted from the loudspeaker is picked up by the microphone and sent back to the far-end speaker. In other words, the far-end speaker hears his or her own voice like an echo. To avoid this phenomenon, most hands-free speakerphones and conferencing systems are equipped with echo cancellers, which decrease or eliminate the echo through signal processing. If the environments at both ends are quiet enough and the echo is not too

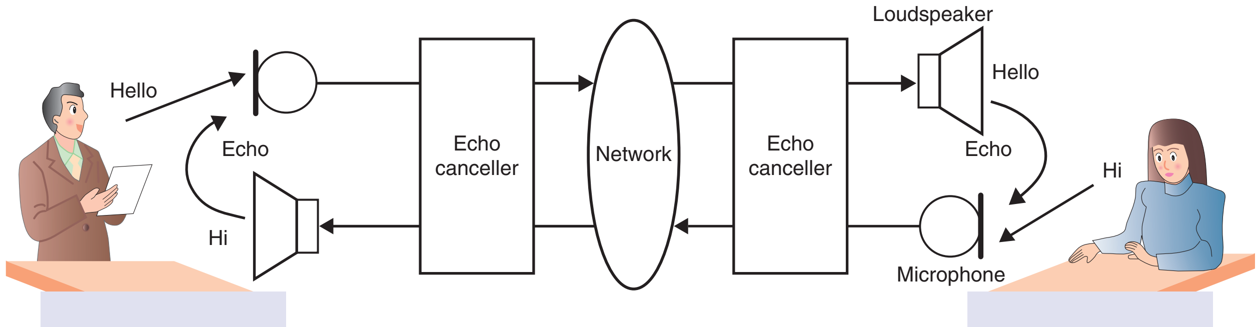

Fig. 5. Hands-free telephone.

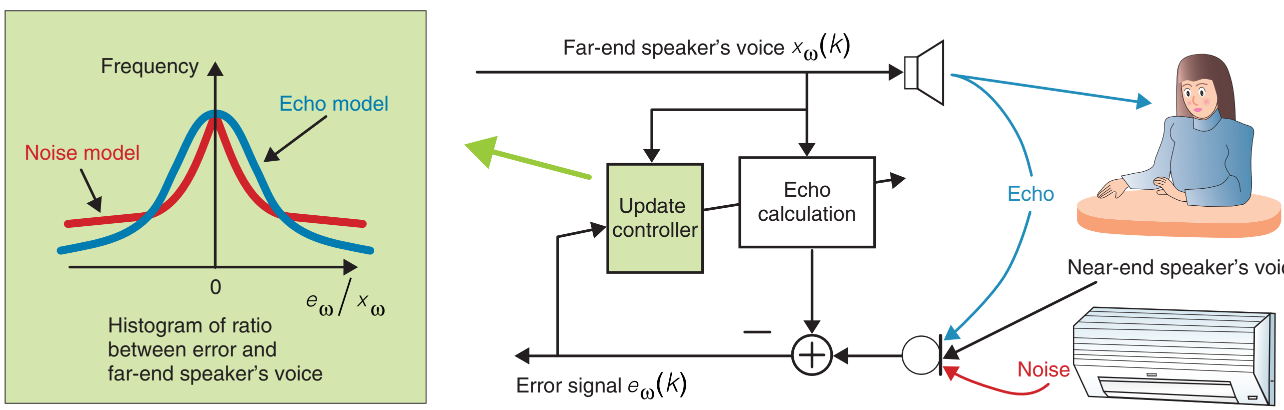

Fig. 6. Noise-robust echo cancellation.

loud, most of these echo cancellers work properly. However, echo cancellation in small devices such as smartphones has two technical problems: the treatment of ambient noise and appropriate hardware design for the device.

Ambient noise deteriorates the echo canceller's performance, especially the adaptive filter's performance. An echo canceller usually has an adaptive filter for estimating the acoustic echo path and calculating the pseudo-echo signal. This estimation is done only when the loudspeaker emits sounds and the near-end speaker does not talk. The microphone picks up only the echo signal, and the adaptive filter estimates the acoustic echo path so that the calculated pseudo-echo can be made equal to the observed signal. If the acoustic echo path is estimated precisely, the echo canceller can eliminate the echo from the sending voice even when the speakers at both ends talk simultaneously (called the double-talk state). However, if the observed signal is contaminated by ambient noise, the estimated echo path has a large amount of error. As a result, the echo canceller cannot eliminate the echo and the remaining echo interferes with communication.

Our new technology, noise-robust echo cancellation (**Fig. 6**), enables echo path estimation by means of an adaptive filter that is less sensitive to ambient noise. This technology uses the ratio of the noise in the microphone signal and controls the update speed of the acoustic echo path. Specifically, the update controller in Fig. 6 calculates the ratio between the far-end speaker signal and the error signal. It then determines whether the noise is dominant within the error signal on the basis of the ratio's statistical distribution. The more noise included in the error signal, the more the update controller decreases the update speed. An echo canceller using this technology cancels echoes three times as much as the conventional echo canceller in a noisy environment, and this performance is suitable for most conditions for hands-free

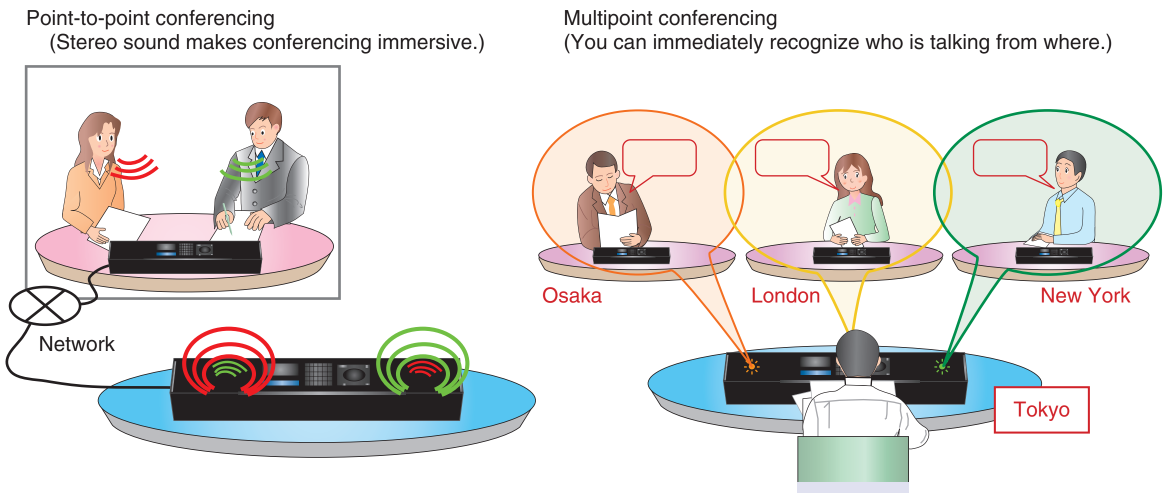

Fig. 7. Typical usages of stereo conference phone.

telephone calls.

The other problem is hardware design. In a small device, the loudspeaker and microphone are located much closer together than in ordinary videoconference systems, and the echo signal is too large to eliminate. Therefore, some ingenuity is needed to make an echo canceller that works well. One solution is to use a directional microphone and locate its low-sensitivity direction toward the loudspeaker in order to avoid picking up echoes. In addition, sound propagating directly along the device's housing to the microphone must be reduced. Adding a rib to strengthen the housing structure, dividing the space between the loudspeaker and microphone, and attaching the microphone to the housing by rubber or sponge mountings, for example, are effective ways to reduce the propagation of vibration from the loudspeaker. It is very difficult to know what the optimal echo cancellation structure is or how to design the hardware; the current design is mainly based on experience. We are considering calculating the optimal structure by computer simulation in the future.

We have now achieved echo cancellation in a small device with performance equivalent to that of an ordinary (large) hands-free telephone. The HIKARI FLET'S phone VP3000 [4], which uses the technologies described above, is now available from NTT EAST and NTT WEST.

5. Stereo conference phone

In recent years, audioconferencing and videoconferencing have been in strong demand in order to cut the cost of business trips and reduce the environmental load. Though audioconferencing (using telephones) is inexpensive, easy to use, and more popular than videoconferencing, it has two drawbacks: it is hard to judge who is talking at the remote site and it is hard to sense the atmosphere at the remote site.

To address these drawbacks, we are developing a new stereo conference phone with 14-kHz stereo sound that corresponds to the quality of FM (frequency modulation) radio. The use of wideband stereo sound makes it easier for people to understand the situation at the remote site.

The stereo conference phone has three features: a 14-kHz audio codec, automatic gain control, and a multimodal display. We extended the 7-kHz codec G.711.1 to 14 kHz. This codec is already standardized as G.711.1 Annex D (G.711.1 SWB). Automatic gain control improves listenability by adjusting voice levels moderately depending on whether the talker is far from or close to the conference phone. The multimodal display uses light emitting diodes and changes their lighting pattern according to the balance between the left and right channels and the levels of stereo sound. This helps users determine who is speaking where at the remote site.

Typical uses of the stereo conference phone are shown in **Fig. 7**. In point-to-point conferencing, FM-quality stereo sound can deliver not only voices but also ambient sound in the remote site vividly. The multimodal display visually assists users in determining who is talking at the remote site. This makes phone conferencing increasingly immersive. In multipoint conferencing, our new stereo conference phone can reproduce sound from each site at a separate position. This overcomes the problem of ordinary multipoint conference phones, which mix voices into monaural sound with the result that you cannot immediately distinguish who is talking from where. Our stereo conference phone makes conferencing more efficient and productive. It can also be used as external hands-free audio equipment with videoconferencing software for a personal computer (PC). Since this audio equipment is self-contained and no tuning is necessary, it will make PC-based high-definition videoconferencing tools more convenient.

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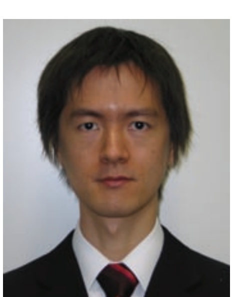

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Visual Softphone: New Ways to Communicate

Koichiro Kanaya, Hironori Ohata, Dai Ando, Machio Moriuchi, Hiroshi Jinzenji, and Jotaro Ikedo†

Abstract

NTT is providing a visual softphone for use on Windows personal computers as a tool for expanding the means of communication. This article provides an overview of it and describes various communication formats that it can provide.

1. Softphone with OAB-J number capability

The visual softphone is a software-based telephone terminal that can run on Windows personal computers (PCs). It is being provided by NTT EAST and NTT WEST free of charge to HIKARI DENWA*1 users of FLET'S HIKARI NEXT broadband service under the name HIKARI Softphone. The initial version of the visual softphone was provided in February 2009 and a content transfer function supporting data connections (enabling data files to be sent and received by a PC) was added in July 2010.

Although Skype and other softphone products are available on the market, achieving a short end-to-end delay has been a difficult problem to solve. At present, only a few softphones (including NTT's visual softphone) can achieve a delay of 150 ms, which is specified as a requirement for using 0AB-J numbers (10-digit numbers starting with 0).

2. Architecture

In general, software on a PC operates by receiving instructions (input) from the user, performing various types of processes, and returning results to the user. In contrast, a softphone operates not only by receiving instructions from the user but also by recognizing and processing an incoming call. A softphone must also support a variety of commercially available external devices such as headsets and cameras. In addition, a softphone by its very nature achieves most of its functions by software, which means that it supports the addition of diverse functions and modification of its operations in a relatively easy manner. To exploit this feature to the maximum, we need to use an architecture that can simplify function addition and specification modification as much as possible without sacrificing overall performance.

With these characteristics taken into account, the visual softphone can be broadly divided into utility, control logic, and graphical user interface (GUI) elements, as shown in **Fig. 1**.

The utility element enables the visual softphone to support various external devices by concealing differences in PC hardware components, and it uniformly supports incoming calls by performing SIP (session initiation protocol) processing and general-purpose processing such as media RTP (Real-time Transport Protocol). The logic element controls all softphone operations and controls input/output according to the state of the software. The GUI element handles input/output with respect to the user.

3. Main functions

The main functions of the visual softphone are listed in **Table 1**. Its functions for videoconferencing, broadband telephony, and content transfer greatly

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*1 Hikari is the Japanese word for light; denwa is the Japanese word for telephone.

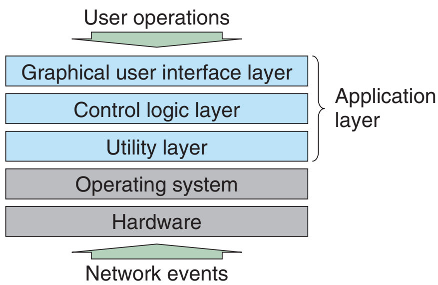
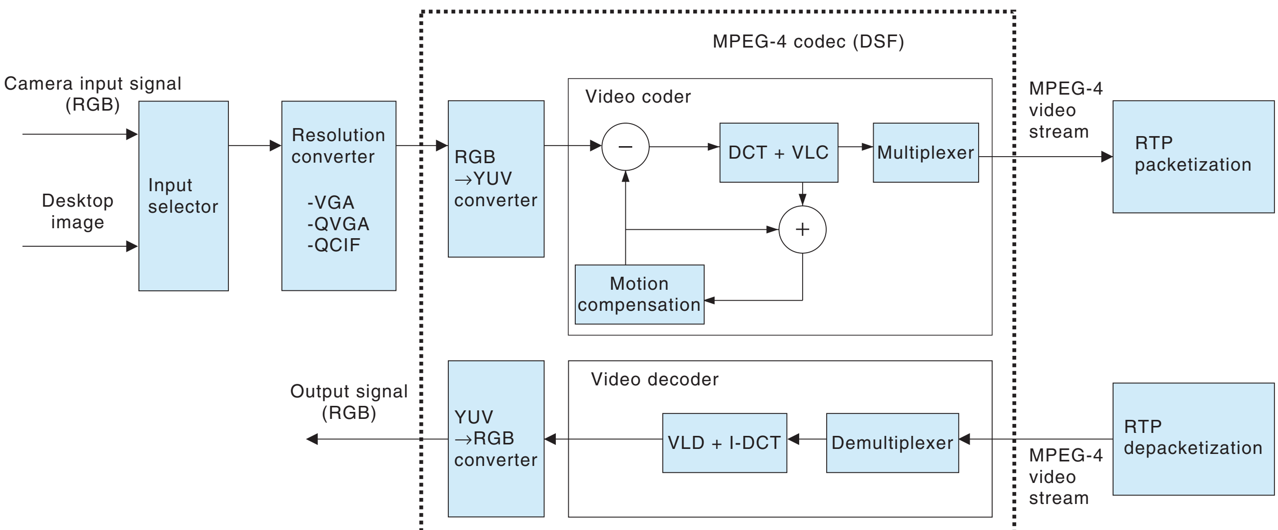


Fig. 1. Architecture overview.

Table 1. Main functions.

Video	VGA (640x480) @ 30 fps: MEPG-4
	QVGA (320x240) @ 15 fps: MPEG-4
	QCIF (176x144) @ 15 fps: MPEG-4
Audio	Wideband (7-kHz) voice: G.711.1 & UEMCLIP
	Narrowband (3.4-kHz) voice: G.711
Digital contents	Simultaneous contents transfer and voice call
	Simultaneous contents transfer and video call
	Individual contents transfer

fps: frames per second



DCT: discrete cosine transform
DSF: DirectShow Filter
I-DCT: inverse DCT
VLC: variable length coding
VLD: variable length decoding

Fig. 2. Visual softphone's video processing module.

exceed conventional telephone functions. These functions can be optimally allocated to suit user needs.

3.1 Video communications function

The configuration of the visual softphone's video communications processing module is shown in **Fig. 2**. This module takes a video signal input from a camera and converts it to a previously set resolution in the resolution-conversion section. It then passes the resulting signal to the video coding section where it undergoes RGB-YUV[*2] conversion and compression/coding. The coding system uses an MPEG-4-based codec developed by NTT Cyberspace Laboratories and achieves high-quality video while suppressing the processing load. Supported video sizes are VGA (640 pixels × 480 lines), QVGA (320 pixels × 240 lines), and QCIF (176 pixels × 144 lines). In general, high-quality videophone communications is provided using VGA video, but QVGA video, which generates a somewhat smaller processing load for coding, is also supported to enable the use of relatively low-performance PCs such as netbook

*2 RGB: red, green, and blue; YUV: color space defined in terms of one luma and two chrominance components.

Table 2. Terminals that can connect to the visual softphone.

	Network	Terminals
Voice call	NGN	HIKARI Softphone
		HIKARI DENWA terminal
	PSTN	Analog terminal
		Mobile phone
		INS voice terminal
		050 number IP phone
		PHS (personal handy-phone system)

INS: Information Network System
NGN: Next Generation Network

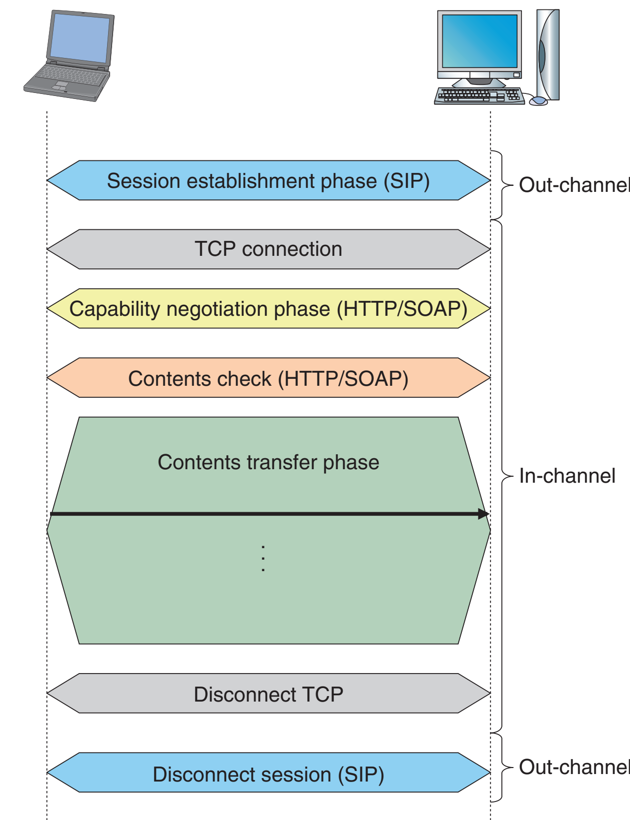


Fig. 3. Example of the contents transfer sequence.

computers. The visual softphone also supports videophone communications with FOMA mobile phones in the same way as the PC Communicator [1], for which QCIF video is used to match the video on the FOMA side.

Although the usual format is for a video signal to be input from a camera and then compressed and sent to the other party's terminal, a desktop sharing function can also be used. In this case, the input-switching section of the video communications processing module switches the input signal from a camera-fed video signal to a rectangular portion of the PC's desktop screen selected by the user. This signal is then compressed and sent to the other party. This configuration enables a screen image to be sent to the other party's terminal during videophone communications without the need to switch applications, which leads to smooth screen sharing.

3.2 Voice communications function

Call control on the visual softphone is performed by SIP, and calls can be made to a variety of PSTN (public switched telephone network) telephone terminals via a home gateway (**Table 2**).

In addition to the G.711 coding system used by the conventional telephone network for speech coding, the visual softphone is also equipped with the UEMCLIP system developed by NTT Cyberspace Laboratories and the G711.1 wideband speech coding system [2] to provide high-quality voice calls with little delay.

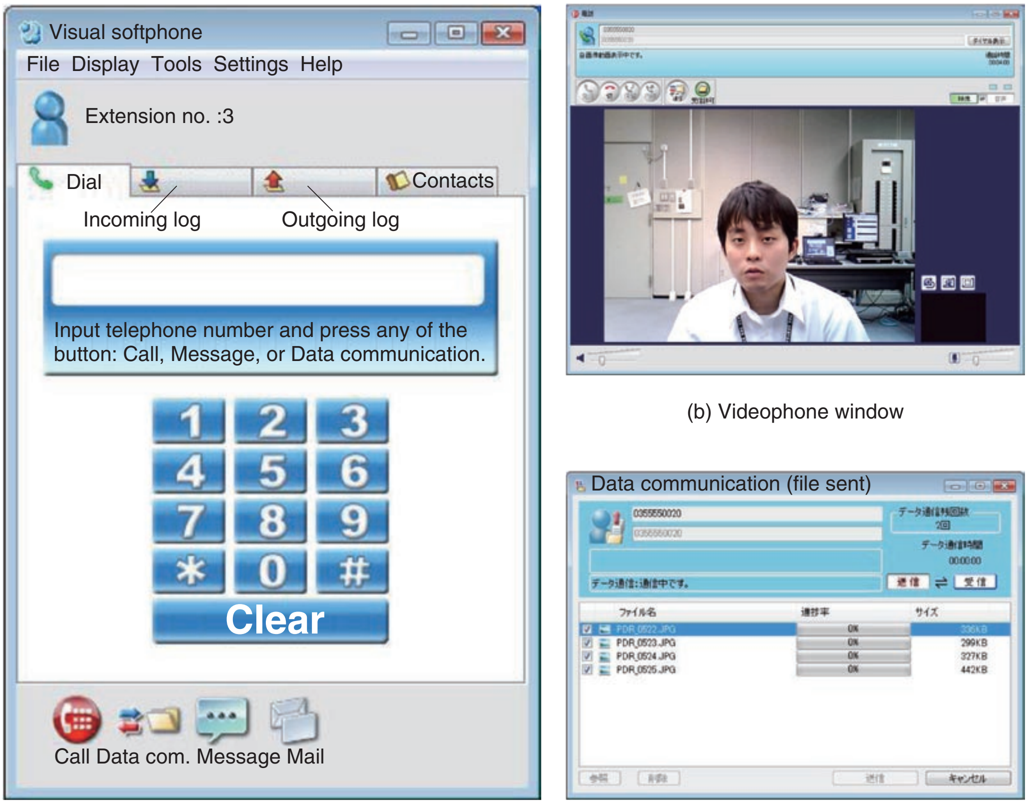
Switching between these speech encoding systems is performed automatically by a negotiation process at the beginning of a call. The user does not need to know what kind of terminal the other party is using.

It is common to incorporate a fixed-size buffer for voice communications that uses Internet protocol (IP) to deal with packet transmission fluctuations and packet loss. However, this buffer processing is directly related to increases in voice delay. In the visual softphone, this problem is dealt with through various measures such as dynamically controlling the buffer size and performing buffer processing frequently with the aim of preventing buffering-related delay as much as possible. The visual softphone also prioritizes the allocation of CPU (central processing unit) resources for voice packet processing to prevent unwanted effects from other applications.

The visual softphone also incorporates a simple echo canceller [3]. This enables the user to connect a microphone and speakers to the PC and perform hands-free calling in an environment where the distance between the microphone and speakers is no greater than 2 m.

3.3 Content transfer function

In addition to video and voice communications,



(a) Main window

(c) Contents transfer window

Fig. 4. Examples of PC windows.

the visual softphone provides a content transfer service that supports the sending and receiving of digital data such as images and documents using the “m=application” media type in SIP/SDP (SDP: session description protocol). This service enables users to send and receive all sorts of digital data in the manner of fax transmission using 0AB-J numbers and even to send data that must be carefully handled such as personal information by utilizing the secure network characteristic of the Next Generation Network (NGN). In short, the content transfer service is expected to stimulate the creation of completely new usage formats.

The content transfer service uses technical specifications for content transfer established by NTT Cyber Solutions Laboratories. The visual softphone can connect to and transfer content to other devices that use the protocol specified in these technical specifications. In particular, the specifications prescribe a method for describing call settings on the basis of SIP/SDP and an in-channel communication protocol using HTTP/SOAP (HTTP: hypertext transfer protocol; SOAP: simple object access protocol) for use after the call has been established.

The content transfer service can be used in both a non-calling state and a calling state. An example of content transfer from a non-calling state is shown in **Fig. 3**. The session is started by a SIP INVITE command specifying m=application. This causes in-channel communications to begin. The softphone on the sending side connects to the softphone on the receiving side by TCP (transmission control protocol). It sends a list of digital content specified by the user (maximum 100 files), transmits the actual digital content, and closes the TCP connection and terminates in-channel communications. The sending softphone sends a SIP BYE command to terminate the session. For content transfer while a call is in progress, the softphone sends a SIP reINVITE command with m=application added to begin in-channel communications. After in-channel communications has been performed, media deletion by reINVITE(port0) is performed to return the communications state to voice/video.

The NGN currently has an upper limit for the number of times that media addition can be performed during a voice/video call, and when m=application media addition is performed at the time of content transfer, a limit can be set for the number of content items that can be transmitted at one time. We therefore considered how to exceed such limits in content transfer by enabling the sending-side softphone to select whether content transfer will continue after one transmission. If it does continue, the user can specify and send new content in the same session by continuing the m=application session.

4. User interface

The GUI is divided into a main window, telephone window, and content transfer window (**Fig. 4**). Information needed at non-calling times, such as contacts and calling history, is all contained in the main window to minimize the display area occupied during non-calling times. At the time of a call, the GUI automatically switches to the telephone window; the procedure for accepting an incoming call was carefully designed to be uncomplicated. The content transfer window enables files for transfer to be selected by drag-and-drop to facilitate intuitive use similar to the operation of other Windows applications.

5. Conclusion

This article outlined NTT's visual softphone, which enables people to communicate using various types of new media on the NGN. Looking beyond the visual softphone's obvious use as a videophone, we intend to propose new ways of communicating using diverse media.

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Software H.264 Encoder Engine for Online Video Delivery Service Cost Reduction

Masaki Kitahara†*, Naoki Ono, and Atsushi Shimizu*

Abstract

This article describes the software H.264 encoder engine that we have developed in NTT Cyber Space Laboratories. It implements technologies for fast encoding, high compression performance, and reduced encoder operation cost, which are all important features for reducing the cost of online video delivery services. In tests, it outperformed conventional encoders. It is also suitable for use in creating content for online video delivery services.

1. Introduction

Video delivery services such as IPTV (Internet protocol television) and video-on-demand have recently gained in popularity. Video generally involves much more data than audio, so the increasing popularity of high-resolution video has led to demands for video encoding. For example, the amount of data used for uncompressed HDTV (high-definition television) video is close to 1 Gbit/s.

Video content delivery over networks requires video encoding technology that can encode video with a good compression ratio, which is defined as the compressed size divided by the original size, so a smaller ratio equates to better performance. To meet this demand, the use of H.264, which is known to achieve twice the compression performance of MPEG-2, has spread widely over the years. H.264 can achieve such high performance because it enables the use of various coding tools, which makes it adaptable to various kinds of video characteristics. However, this means that an H.264 encoder must adaptively select coding tools from a large number of candidates according to the characteristics of the input video in order to achieve a good compression ratio. In other words, there is a trade-off between compression performance and encoding time: H.264 encoders tend to need long encoding times. To reduce the service cost of online video delivery services, it is necessary to encode video with high compression performance so that the bitrate of the compressed video is low in order to reduce storage costs. Therefore, to reduce the encoder operation cost in such services, fast encoding is needed. The basic functionalities and system requirements are listed in **Table 1**.

NTT Cyber Space Laboratories has a long history of developing codec LSIs (large-scale integrated circuits) and equipment. We utilized know-how acquired from those past developments and developed technologies for fast encoding, high compression performance, and encoder operation cost reduction, which are all important for reducing the cost of online video

Table 1. Basic functionalities and system requirements.

Encoding format	MPEG-4 part 10 AVC/ITU-T H.264	
Profiles	Main Profile, High Profile, High 4:2:2 Profile	
Image size	CIF (352 x 288) to 4K x 2K (3840 x 2160)	
System requirements	CPU	SSE4.1 or above
	OS	MS-Windows XP SP3
	Library	Microsoft C/C++ Runtime Library

CIF: common intermediate format
OS: operating system

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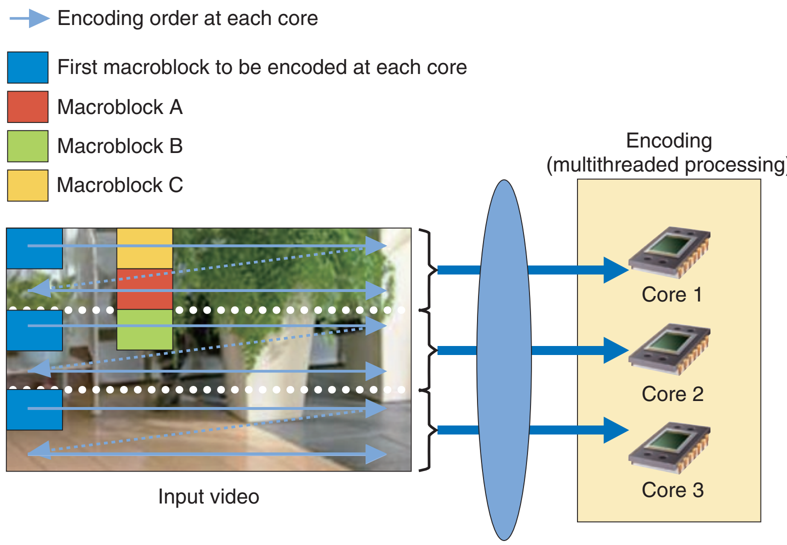

Fig. 1. Conventional multithreaded encoding.

delivery services. These technologies are described below and compared with other encoders.

2. Technology for fast encoding

We have implemented a multithreaded encoding method that utilizes multicore central processing units (CPUs), which are common in recent personal computers (PCs), with negligible degradation in compression performance.

Conventionally, multithreaded encoding has been achieved by dividing images in the input video into sub-images and encoding the sub-images independently thread by thread. Such a method is illustrated in **Fig. 1**. In H.264, images in the input video are encoded in units of macroblocks, which are blocks of pixels that consist of 16 pixels in the horizontal direction and 16 pixels in the vertical direction; when a macroblock is encoded, information about how adjacent already-encoded macroblocks were encoded is utilized in order to achieve high compression performance. Furthermore, encoding begins with the macroblock at the top left of the image (or the sub-image if conventional multithreaded encoding is used) and each macroblock in the top horizontal macroblock line is encoded from left to right. When the top right macroblock is encoded, macroblocks in the horizontal macroblock line below are encoded from left to right, and the rest of the horizontal macroblock line is processed in the same way. When the encoding method described in Fig. 1 is used, at the moment core 2 is encoding macroblock B, core 1 is encoding macroblock C, assuming that the encoding speed is the same for all the cores. This means that macroblock A, which is one of the macroblocks adjacent to macroblock B, has not yet been encoded by core 1, so its encoding information is not yet available for use in encoding macroblock B; this results in low compression performance. Since the number of such cases increases with the number of divided images (faster encoding), it is not possible to achieve a significant speedup in encoding in the conventional method without significant degradation in compression performance. Thus, even if the PC had many cores, it would not possible to utilize its full potential with the conventional method.

In the newly developed software H.264 encoder engine, we have implemented a new method created by applying know-how acquired during encoder LSI development. In this method, the encoding of macroblocks is pipelined, so that the number of threads can be increased with negligible loss in compression performance. Thus, it is possible to utilize the full potential of newer multicore CPUs designed for PCs. As a result, our new H.264 encoder engine can encode at twice the speed of our conventional one. Furthermore, even faster encoding will be possible in the future by using PCs with more cores than those on the market at present.

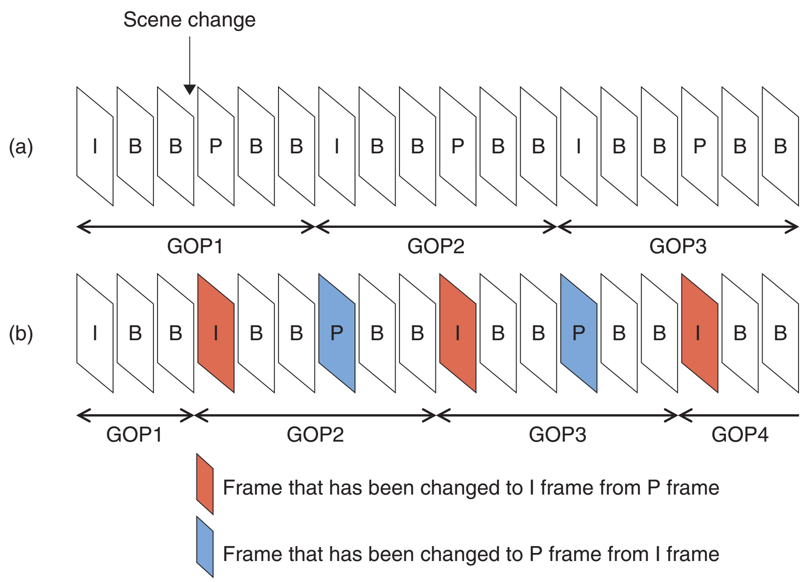

Fig. 2. Adaptation of GOPs to scene change.

3. Technology for high compression performance

We have implemented a method that adapts the structure of groups of pictures (GOPs) to scene changes to achieve higher compression performance. In H.264, frames in the input video are grouped together to construct GOPs. Furthermore, each frame can be one of three types: intra-coded (I), predictive-coded (P), or bidirectionally predictive-coded (B). I frames are encoded solely using information inside the frame to be encoded without reference to other frames, P frames are encoded using past I or P frames and are also used as a reference for further prediction, and B frames are encoded using both past and future frames. An example where each GOP is of equal length (6 frames), and there are 2 B frames between an I frame and a P frame or between two P frames is shown in **Fig. 2(a)**. As shown here, it is common to use I frames for the first frame in a GOP to enable random access.

Since P and B frames use past or future frames for encoding, when a scene change occurs in such frames, the compression performance degrades significantly because there is no correlation between the frame to be encoded and the frame that is used for encoding. The conventional method for avoiding this was to change the frame type of such a frame to an I frame, leaving the GOP structure basically unchanged. However, such a simple solution leads to an increase in the number of I frames. Since the compression performance of I frames is low, this is not an optimal solution for coping with scene changes.

An example of the method implemented in the new encoder engine is shown in **Fig. 2(b)**. When a scene change occurs, a new GOP is created from that frame. With this method, the number of I frames can be smaller than in the conventional method. Furthermore, since the GOP length can be limited, this method is applicable to the case where the GOP length needs to be made shorter than a particular value, such as in IPTV system specifications.

When this method is used together with the fast encoding technology described in section 2, the bitrate of compressed video can be 20% smaller compared with our conventional software H.264 encoder engine at the same encoding speed.

4. Technology for encoder operation cost reduction

In general, when the bitrate of a long section of compressed video exceeds the target bitrate, it is impossible to decode the video within the specified frame rate because this section of compressed video takes longer to deliver over a network that has bandwidth equal to the target bitrate. Such cases occur when the video contains a section of content that is complex and difficult to compress; an example is a scene with water splashing. In such cases, the person

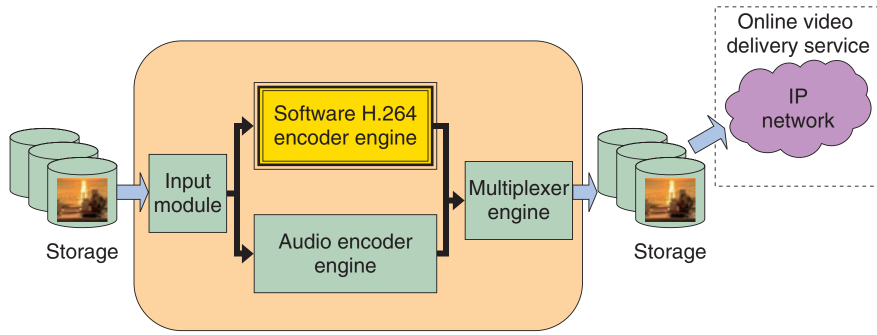

Fig. 3. Example of an audio-visual encoder application.

Table 2. Comparison with other encoders.

	Competitor's encoder	Our conventional encoder engine	Our newly developed encoder engine	Remarks
Compressed bitrate*1	7 Mbit/s	7 Mbit/s	5.6 Mbit/s	-
Encoding time*2	x3.3	x5.2	x2.6	Encoder processing time at the same bitrate
Automatic retry of encoding	No	No	Yes	-
Encoding in 4:2:2 format video	Yes	No	Yes	-

*1 Bitrate able to provide the video quality of digital terrestrial television broadcasting in Japan
*2 Encoder processing time relative to the actual video length when HDTV video was encoded

operating the encoder must encode the video again using different encoder parameters to decrease the bitrate of the complex parts. One problem with this is that the changed encoder parameters can influence other parts of the video, resulting in the video having degraded video quality. Another problem is that since encoding with different parameters can occur multiple times and the encoder operator must check the quality of the video each time, the encoder operation cost is higher.

In our new H.264 encoder engine, we have implemented a method that automatically detects anomalies during encoding and automatically encodes such parts with different encoder parameters multiple times. This method avoids quality degradation of the whole video and achieves a lower encoder operation cost.

5. Use of the H.264 encoder engine for application development

The H.264 encoder engine that we have developed is solely for encoding images. To develop an application that can encode video content containing both images and sound, one needs other components such as an audio encoder engine and a multiplexer engine. An example of the architecture of such an audio-visual encoder application is shown in **Fig. 3**.

6. Comparison of performance and functionalities

The performance and functionalities of our new software H.264 encoder engine are compared with those of our conventional encoder engine and a competitor's encoder engine in **Table 2**. Our new encoder achieved a 20% reduction in the bitrate of compressed video at the same encoding speed. It supports the 4:2:2 format, which is important for professional use, and it also has functionalities that are not implemented in competitors' encoder engines such as automatic retry of encoding (described in section 5).

7. Conclusion

Our new software H.264 encoder engine achieves fast encoding, high compression performance, and a lower encoder operation cost, which are important features for reducing the cost of online video delivery services. It is therefore highly suitable for use in creating content for online video delivery services. Our future work includes developing encoder applications based on the software H.264 encoder engine and extending the technologies described here for use in other kinds of services.

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Network Processing Technology for Terminals Enabling High-quality Services

Yukikuni Nishida† *and Keiichi Koike*

Abstract

This article describes the RENA-SoC, which is a network processor that provides basic functionalities required for a home or SOHO (small office home office) network gateway in the fiber-to-the-home (FTTH) broadband era (RENA: routing engine for Next Generation Network access, SoC: system on a chip). Network processing is performed by a packet processing engine, and the application processing load is distributed over two microprocessors to provide the user with high-quality stable services.

1. Introduction

Managed networks such as the Next Generation Network (NGN) can provide high-quality, secure networking for triple-play services that provide voice and video communications, video delivery, and Internet access. Personal computers (PCs) as well as televisions (TVs) and various sensors will be connected to home networks, enabling operations such as TV program reservation or remote monitoring of sensors. Many other services are likely in the future even from home appliances such as refrigerators and microwave ovens connected to the network.

The gateway is very important for connecting the managed network with home networks and small office home office (SOHO) networks, while also protecting the network and the access points for receiving various services. As shown in **Fig. 1**, the gateway must provide many functions such as security, quality control, and service enhancement, as well as the performance to process packets of Gigabit Ethernet signals.

The main functions of the gateway can be classified into network functions and application functions. Application functions must support voice over Internet protocol (VoIP), a basic service of managed networks, and may require the OSGi (Open Services Gateway initiative) framework for managing application software, so that application software can be added and deleted on demand. OSGi runs on the Java VM (virtual machine), which avoids the need to take into consideration differences in platforms, reduces the need to develop application software for each platform (called *bundled* software in Java VM terminology), and makes development inexpensive. In this way, users are expected to be able to get applications at a reasonable cost.

At NTT Laboratories, we are promoting open innovation and have developed the OSGi Service Aggregation Platform (OSAP). This is a base technology that extends OSGi, allowing bundled software from service providers for instance, software that lets the service providers control home appliance, to be downloaded and run on the gateway.

If microprocessor-intensive application software is executed in a multitasking environment, the load on the central processing unit (CPU) will increase and the performance of packet processing, which is a basic gateway function, could decrease and lead to longer packet delays. This could result in a decrease in VoIP quality. To avoid this sort of quality of service (QoS) degradation, we designed an architecture that distributes processing over three processing engines:

- a newly developed gigabit-per-second packet processing engine,

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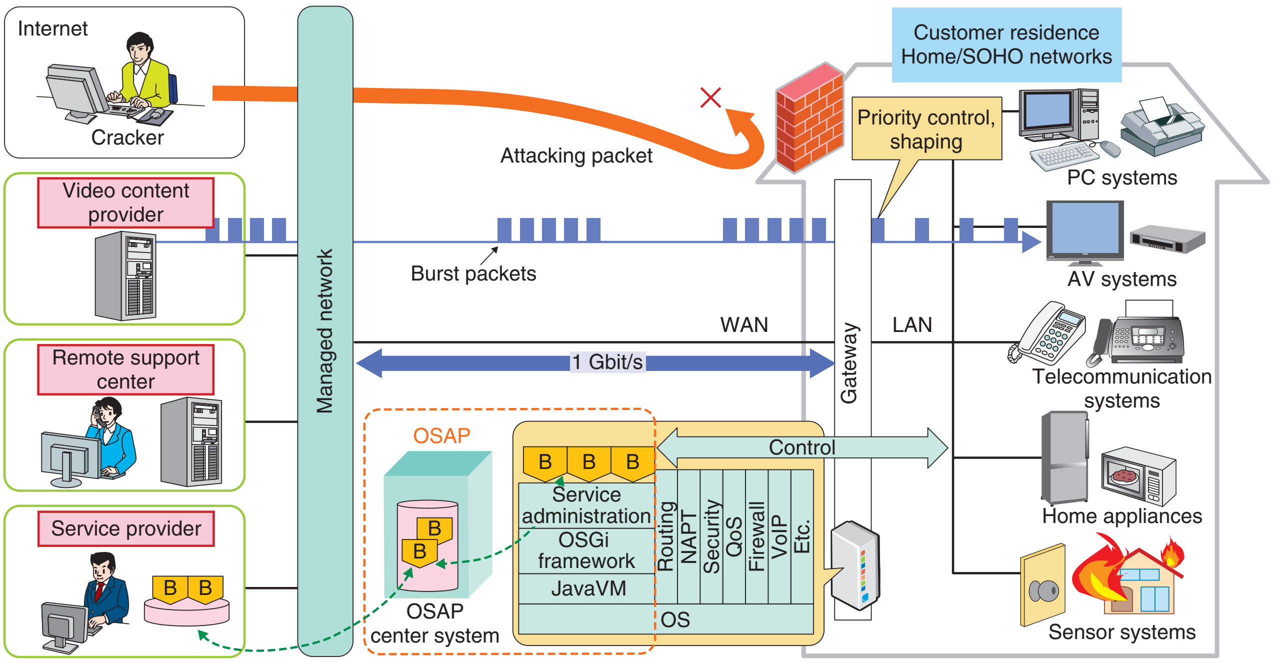

AV: audio-visual
B: software bundle
NAPT: network address port transIstion
OS: operating system
PC: personal computer
WAN: wide area network

Fig. 1. Home/SOHO networks.

- a CPU for processing primary services such as VoIP,
- a CPU for processing enhanced functions such as OSGi bundles.

Gateways are also required to provide interfaces for devices such as wireless local area network (LAN) cards and USB (universal serial bus) hard disk drives, and a LAN switch (LAN-SW) in order to connect various pieces of network equipment. This can lead to a large number of components and result in increased power consumption and cost, so to reduce power consumption and cost, we developed the RENA-SoC (RENA: routing engine for Next Generation Network access [1], SoC: system on a chip), which is a network processor that incorporates a packet processing engine, two CPUs, PCI (peripheral component interconnect) and USB interfaces, and a LAN-SW on a single chip (**Fig. 2**).

2. Packet processing engine

The packet processing engine, which contributes greatly to the packet processing performance of the RENA-SoC, is composed of an IPsec block, an IP switch (IP-SW) block, a LAN switch (LAN-SW) block, and a packet scheduler. The IPsec block handles packet encapsulation for virtual private network communications, high-speed encryption, decryption, and authentication, the IP-SW block handles packet transmission as required for each connected party, including firewall and NAPT (network address port translation) functions, and the packet scheduler controls the QoS. These functional blocks are implemented in hardware, and they achieve processing performance of 1 Gbit/s in each direction (uplink and downlink). To confirm the performance of the packet processing engine, we compared the throughput of an off-the-shelf software-based router and a router implemented using the RENA-SoC for various frame lengths and NAPT table entry numbers. The results are shown in **Fig. 3**. The software-based router showed throughput dependent on the packet length and the number of simultaneous flows for NAPTs since these affect the software load. However, the RENA-SoC was not affected by these factors, and achieved 1-Gbit/s bidirectional packet processing. This packet processing engine enables low-delay, high-quality communications, even under high-load

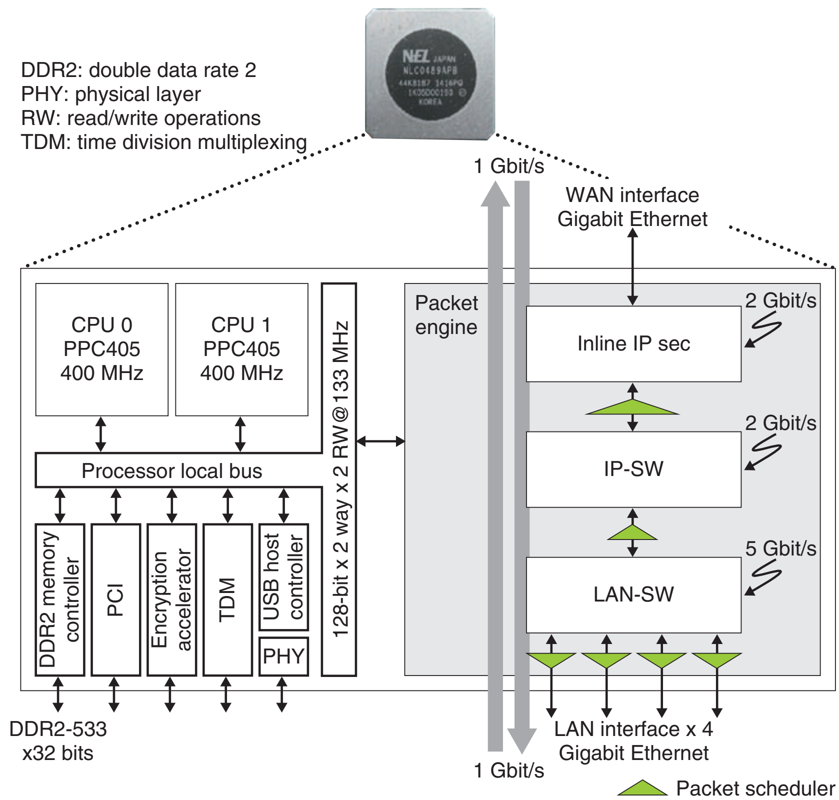

Fig. 2. RENA-SoC block diagram.

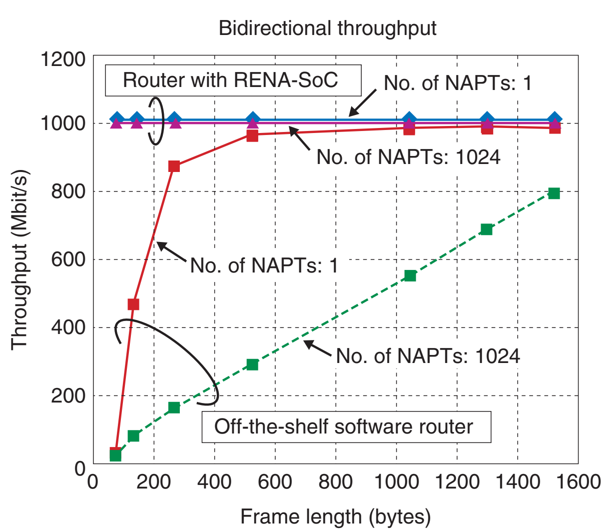

Fig. 3. Packet transmission performance.

conditions of these sorts, by processing packets at the maximum interface speed of 1-Gbit/s.

The packet processing engine can also support new services through the following added functions.

- Communication quality monitoring function (packet copy function)
- Packet processing support function (improves the flexibility of hardware processing)

These functions are explained in detail below.

3. Communications quality monitoring function

When packets are discarded along the communications path, which causes a QoS decrease in services such as video delivery, it is necessary to detect whether the packets were discarded on the WAN (wide area network) segment or LAN segment. Maintenance costs can be reduced if the detection is not performed by a dispatched worker but by a telemetering function. It is currently done by placing monitoring equipment on the home network segment and dispatching a worker to the location to check the log information, which involves a significant amount of labor. Thus, there is demand for remote realtime monitoring of

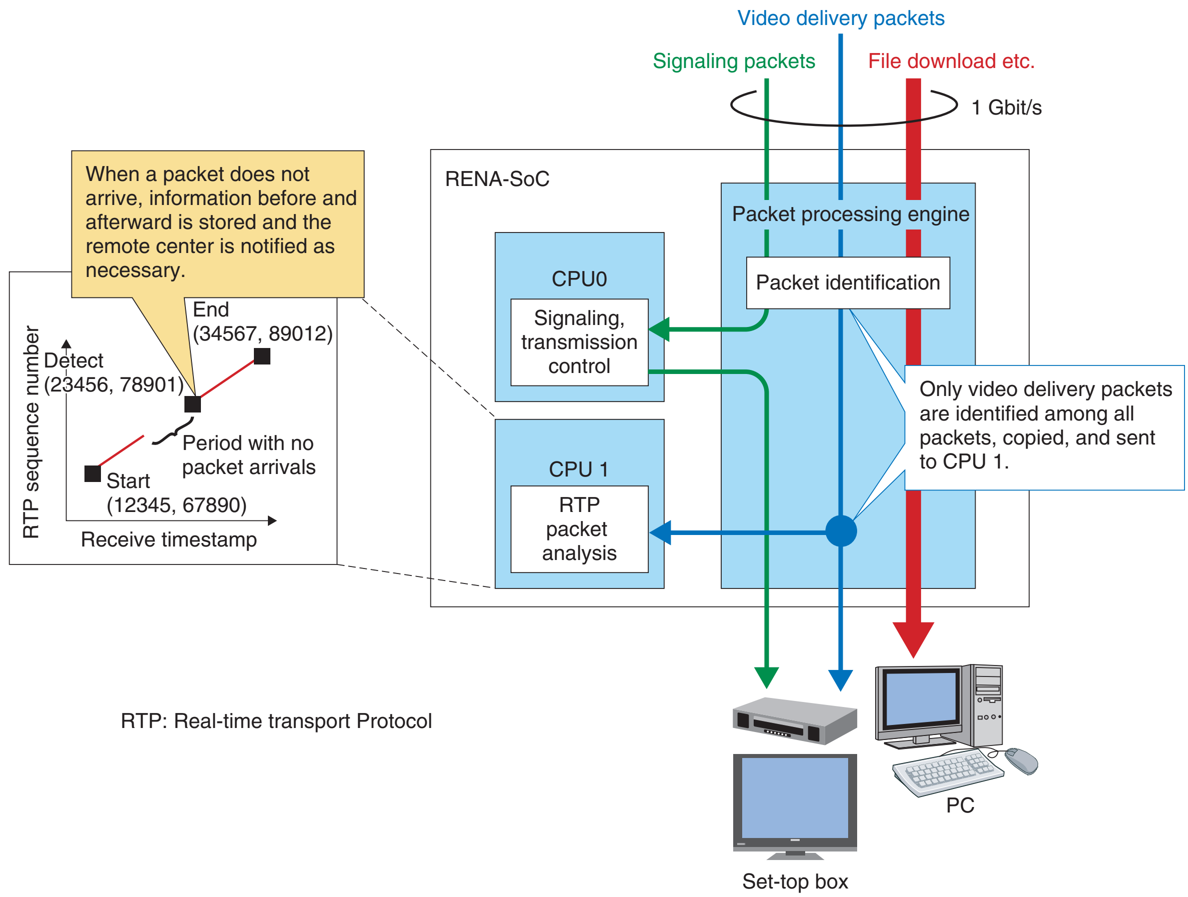

Fig. 4. Quality management for video delivery packets.

communications quality in gateway equipment.

To improve transmission performance when software is used to process packets, often only the headers are read, which avoids the need to read and write entire packets. When packets are captured to monitor quality, entire packets must usually be copied, separated from the packet transmission stream, and passed to the packet capture application. Continuous packet copying makes the processing load heavy and can cause packet transmission performance to decrease significantly, so even if video packets, for example, would have been transmitted correctly, communications quality can be degraded and packet capturing may fail. This can result in faults and errors being detected even when packets could have been received normally.

For these reasons, we have implemented a packet-copy function in hardware that allows communications to continue while copied packets are processed by the CPU and allows realtime packet capturing to be done without decreasing communications quality. Specifically, the packet processing engine identifies packets to be captured and copies them as shown in **Fig. 4**. Even with packets flowing at 1 Gbit/s, specific packets can be copied to the CPU without affecting the packets being transmitted. The CPU can capture specially selected packets, so quality monitoring can be done without causing further packet loss. Figure 4 shows an example of a function that monitors sequence numbers and timestamps of video delivery packets. It also shows that information before and after the event is saved if a sequence number is not continuous or if packets do not arrive for a period of time. This information is reported to a remote center when necessary, which ensures that the center is notified of decreases in service quality.

The above structures enable immediate remote identification of where a network fault occurred from a remote location.

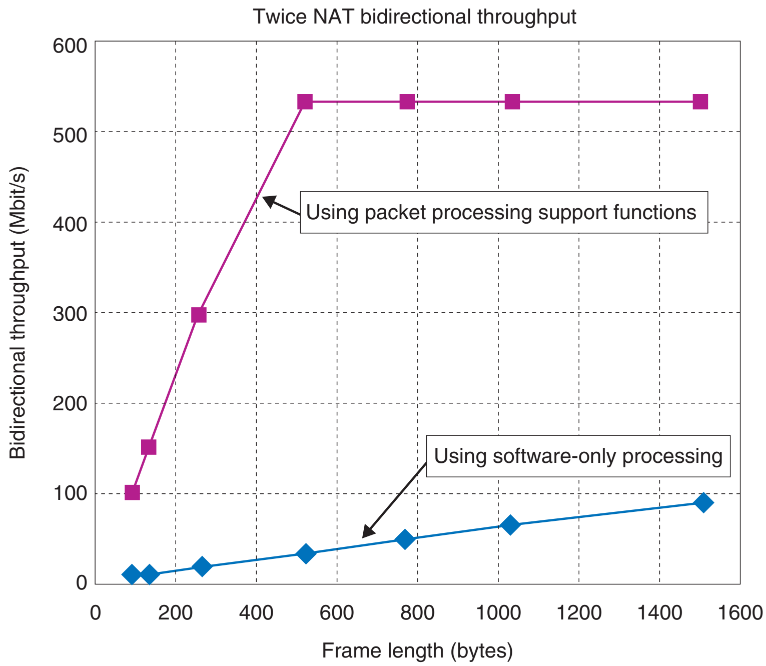

Fig. 5. Performance of packet processing support functions.

4. Packet processing support functions

Packet processing engines are generally composed mainly of hardware, so they have extremely high-performance packet processing; on the other hand, they are inflexible in terms of support for new functions. If the hardware cannot handle packets, even if it is because of a trivial modification to the packets, all of the packets must be processed by software, which causes a dramatic degradation in performance. To resolve this problem, the RENA-SoC implements packet processing support functions that allow the CPU to directly read and write packet data being processed by the packet processing engine. This allows higher performance than is possible with software-only processing.

When packets are processed using only software, the packet processing engine usually passes the entire packet to the CPU, which performs software-based network processing, and then passes the packet back to the packet processing engine. This uses up a lot of processor resources for the packet processing. By enabling the CPU to directly handle packets being processed by the packet processing engine, we were able to reduce the transfer times between the CPU and packet engine and omit most of the network processing.

As an example, for the case of supporting twice-NAT (RFC 2663), in which IPv4 addresses of both source and destination are rewritten, we compared software-only processing and the use of packet processing support functions (**Fig. 5**). For software-only processing, the maximum possible transmission rate was 90 Mbit/s; however, with packet processing support functions, we confirmed transmission speeds over 500 Mbit/s.

5. Application protection

By opening up the system to OSGi bundles, as described earlier, we enable various user applications to be executed on the gateway. To prevent these applications from affecting the operation of basic services such as VoIP, the RENA-SoC uses asymmetric multiprocessing, with an independent OS (operating system) operating on each CPU. Having two OSs running on the two CPUs increases the amount of memory required; however, even if, for example, CPU1 stops responding owing to an inadequately tested OSGi bundle, CPU0 can reset only CPU1, which can restart while CPU0 and the packet engine continue to operate independently without being affected. This makes system operation more stable and keeps primary services alive.

6. Summary

We have developed the RENA-SoC, which enables the operation of high-performance systems through its triple-processor architecture, consisting of a packet engine capable of 1-Gbit/s bidirectional packet transmission with support for IPsec, firewalls, and quality control; a CPU for handling basic managed-network services like VoIP; and a CPU for handling enhanced applications such as OSGi bundles or quality monitoring.

In the future we will continue research and development toward reducing the device's power consumption and providing support for various services.

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Regular Articles

Cabling Technologies Providing More Optical Cabling Potential in Multi-dwelling Unit Buildings

Hayato Minami[†]*, Kensei Shiraishi, Tadashi Sasaki, Kazutoshi Takamizawa, Tetsuhiro Numata, and Osamu Inoue*

Abstract

For optical access services to multi-dwelling units (MDUs), we have developed compact optical splitter modules that provide greater installation flexibility and indoor optical cabling that can be installed on the building's exterior walls. They overcome problems encountered in providing optical fiber cabling systems to MDUs and installing splitter modules in the limited space available in main-distributor-frame boxes as well as other difficulties caused by a lack of space in conduits or the absence of conduits.

1. Introduction

In June 2010, the number of fiber-to-the-home (FTTH) contracts in Japan surpassed 18 million, heralding the age of ubiquitous optical fiber access. However, since roughly 40% of all households in Japan live in multi-dwelling units (MDUs), such as apartment buildings and condominiums [1], there is growing demand for more adaptable optical cabling systems to provide fiber to households in these sorts of dwellings.

In contrast to FTTH cabling schemes for detached houses, which consist of a 4-branch optical splitter in the central office and 8-branch splitter module parts (SPs) installed on the user side (supporting 32 subscribers), schemes for MDUs and office buildings use 32-branch SPs (or a cascade of one 4-branch SP followed by four 8-branch SPs). To date, thin low-friction indoor optical fiber [2] that enables more cables to be installed in conduits and SPs that can be installed in the empty spaces scattered throughout customer buildings (e.g., telephone conduits and main distribution frame (MDF) boxes) [3] have been developed and deployed to expand the range of available cabling systems. They have helped to reduce the number of MDU buildings unsuitable for optical fiber cabling.

However, as indicated in **Fig. 1**, many medium-sized and large MDU buildings have a lack of empty space available in existing MDF boxes etc., and the metallic nature of the hardware already installed means that the conventional SPs cannot be installed easily, and many small MDUs have no conduits or MDF boxes. These conditions make it difficult to provide optical-fiber-based services to such buildings. To tackle these problems, we have developed (1) technologies for installing SPs in MDF boxes in medium-sized MDU buildings, (2) cabling that can be installed on the exterior walls of small MDU buildings, and (3) technologies for installing SPs in large MDU buildings.

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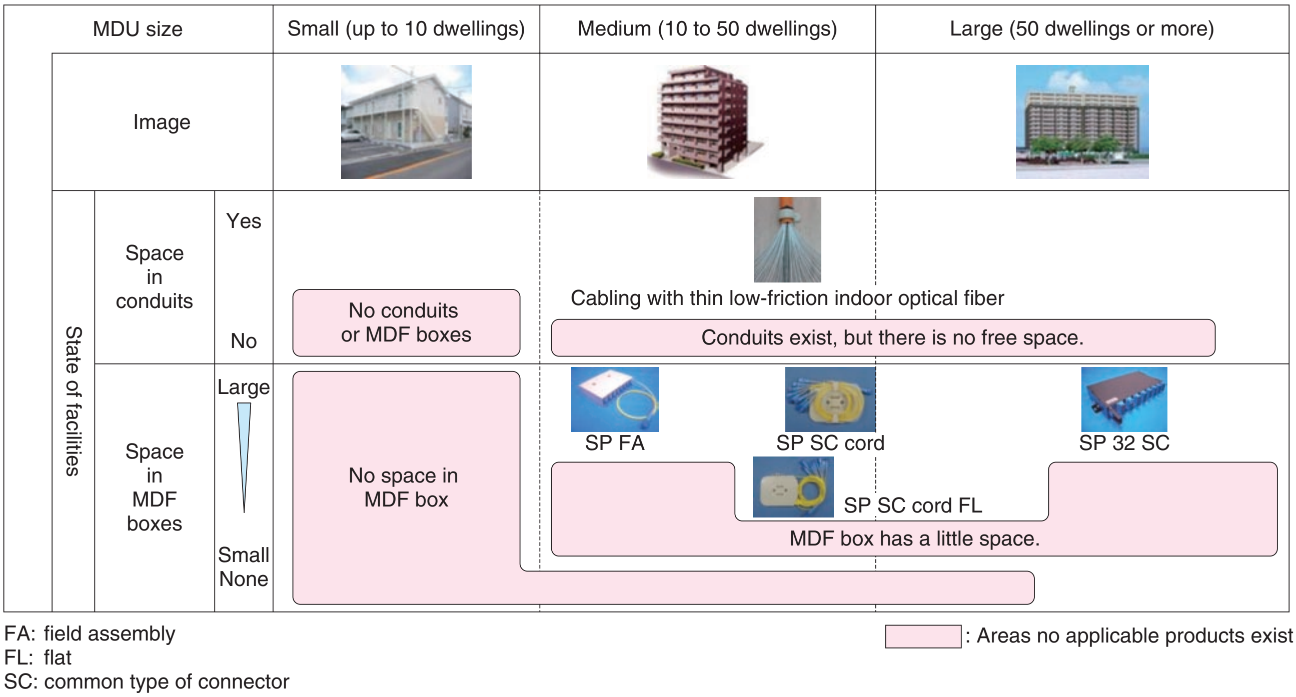

Fig. 1. Current state of hardware applications.

2. SP installation technologies for medium-sized MDU buildings

Since the SP hardware needs to be small enough to fit into the limited space available in existing MDF boxes, we developed a range of smaller SPs that includes both an ultra-compact SP that can be installed in very small MDF boxes and compact SPs designed to provide better workability and allow more flexible installation methods (**Fig. 2**). These are described below.

2.1 Ultra-compact SP

For SP installation in very small spaces, we further miniaturized the existing SP (called the SC cord FL (flat), shown in Fig. 1) [2]. An overview of the new SP (SC cord S (small/stick)) is shown in Fig. 2. By devising a way to fit the splitter section inside the casing, we reduced the space required for installation by 20%. We also developed multiple alignment members that can be fitted to individual cores, which protect the fiber core connections used for feeding the cord to subscriber residences.

2.2 Compact SPs

In dealing with the increasingly diverse cabling forms that result from cabling schemes in optical connection systems used for feeding single-core cable (single-drop optical fibers etc.), it is more efficient to assemble a cable sheath fixing connector (plug type) and terminate it directly at the SP. Moreover, there is a growing need for more versatile attachment methods that do not rely on only ordinary screws, but also enable attachment to metal or plastic. Our new compact SPs (B and C types) are overviewed in Fig. 2. We reduced the space required to maneuver the hardware and install an SP by including input and output sockets and by eliminating the cord section. Unlike conventional equipment where the connectors are secured with screws, these SPs use magnetic attachment to the SP and hinged SP mechanisms. The new design enables much easier detachment and attachment and greatly simplifies the connection work required inside small MDF boxes.

3. External wall cabling technology for small MDUs

Some small condominiums do not have an MDF box or conduits available for cabling subscribers desiring optical fiber services. In this situation, a commercially available box is usually attached to the

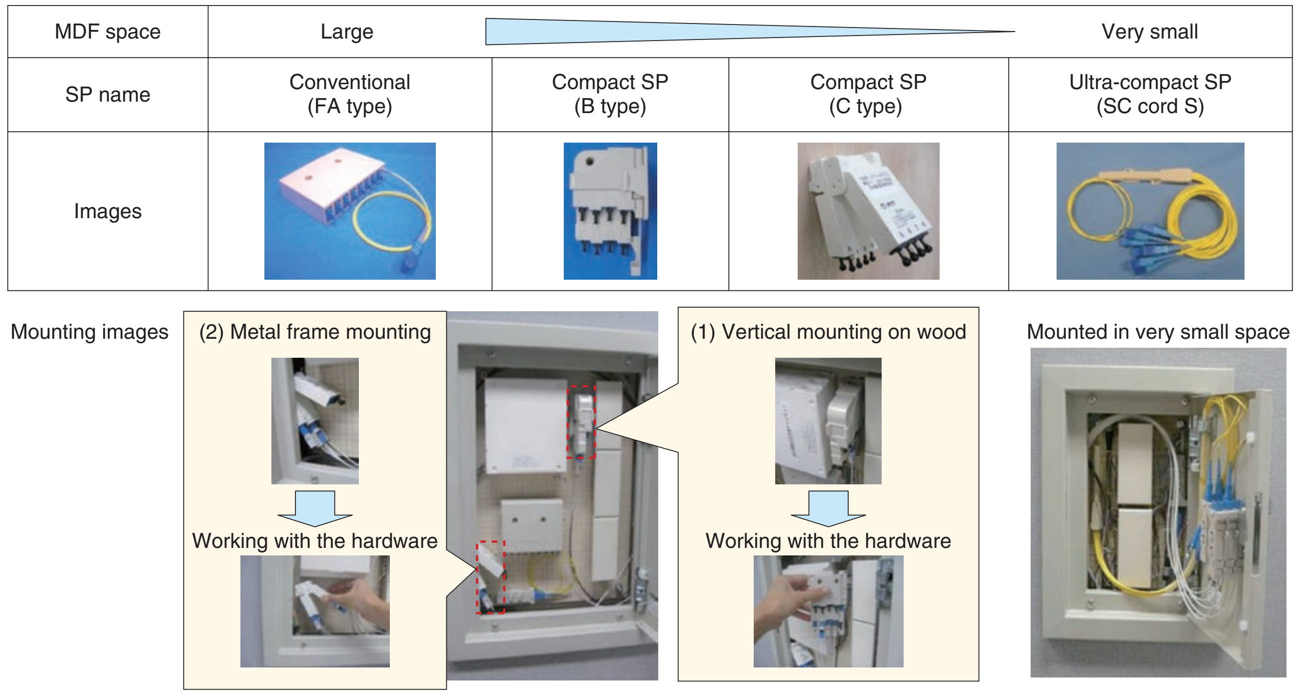

Fig. 2. Technologies for mounting SPs in MDF boxes in medium-sized MDUs.

exterior of the building and the SP is installed inside it. Cabling enters the subscriber's home via the balcony, but this means that work is performed when each individual residence applies for subscription, and negotiations with neighbors are required for permission to work on communal walls (**Fig. 3(a)**).

To address these problems, we have developed a compact box-type SP for attachment to building exteriors and indoor optical fiber bundles that eliminate the need for repeated negotiations with neighbors after the initial application for work.

An overview of the hardware is shown in **Fig. 3(b)**. We used the casing for two existing compact attachable boxes (PD and SS types) and it opens and closes in the same way. We designed the units to be resistant to ultraviolet (UV) light and water in order to be suitable for external installation. The indoor optical fiber bundle is a cable consisting of 8 UV-resistant indoor optical fibers pre-twisted helically with suspension members. When an optical fiber is fed into a subscriber residence, one of these optical fiber cores is selected and connected to the sheath grip connector (FA (field assembly) connector [3]) and then laid.

These systems can reduce the need for problematic negotiations with neighbors about service connection work.

4. SP installation technologies for large MDUs

In cases where there are a large number of prospective users in a large MDU, it is more efficient to batch-install 32-branch SPs rather than the cascades of 4- and 8-branch SPs usually installed in small and medium-sized MDUs. For large MDUs, in addition to installations in the empty space in MDF boxes, there is also demand for 19-inch rack-mounted SPs. Conventional 32-drop SP units are the same height as one 19-inch rack unit (1U), but taking into consideration the space required for maneuvering the input and output cabling, each one needs 3U of free space. Moreover, the installation of more than one conventional 32-drop SP (e.g., to handle 64 or more terminals) in a 19-inch rack requires the use of more than 3U. Therefore, it would be desirable to have more compact hardware that can be installed in a standard 19-inch rack and can make better use of the space available.

To ensure enough work space, we reconsidered the pitch between the usual connectors and chose to use a modular scheme that enables the connector sections to be moved horizontally in and out of the chassis. Simply sliding the required section to insert a connector ensures the same level of workability as with the

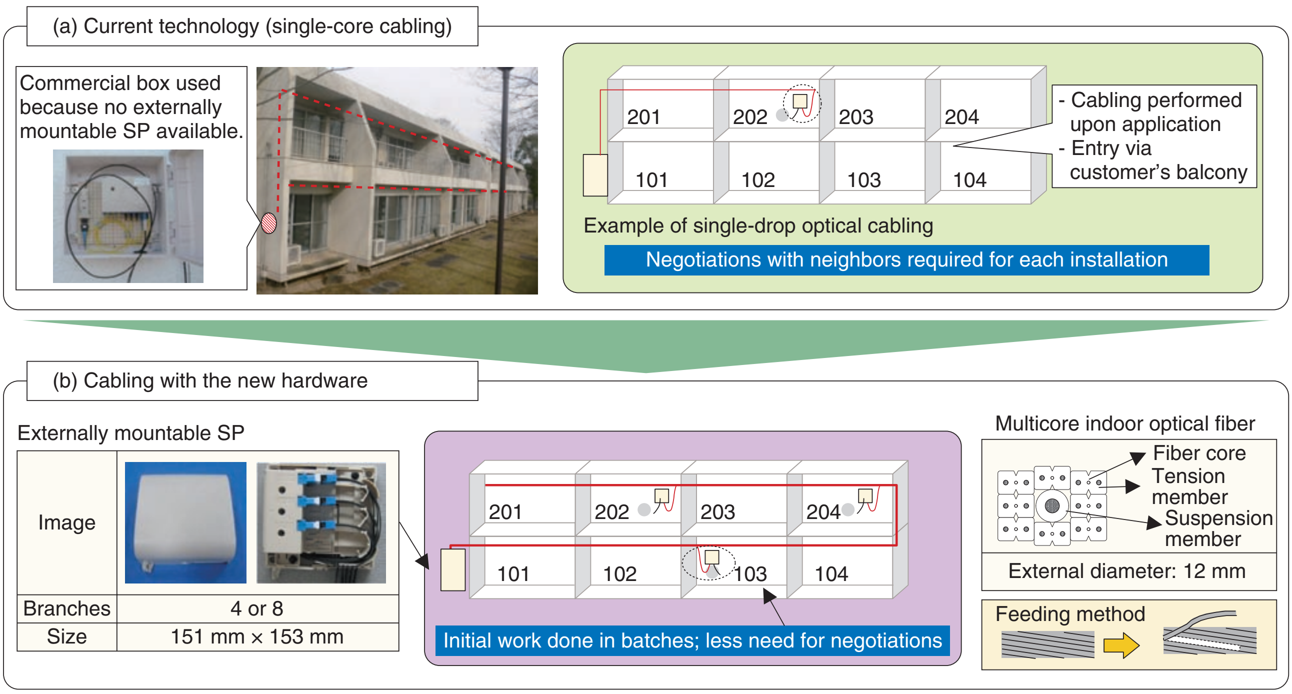

Fig. 3. Comparison of conventional exterior cabling and the new system for small MDUs.

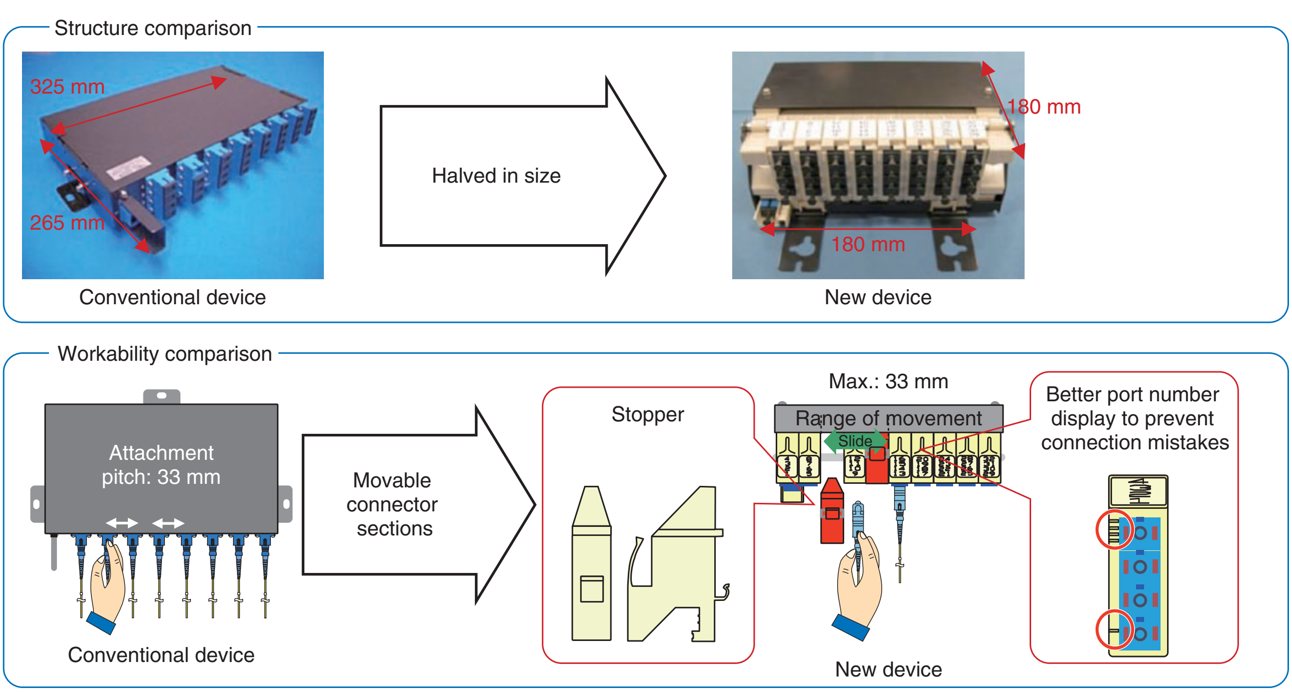

Fig. 4. Comparison of conventional and new devices (for 32 SPs).

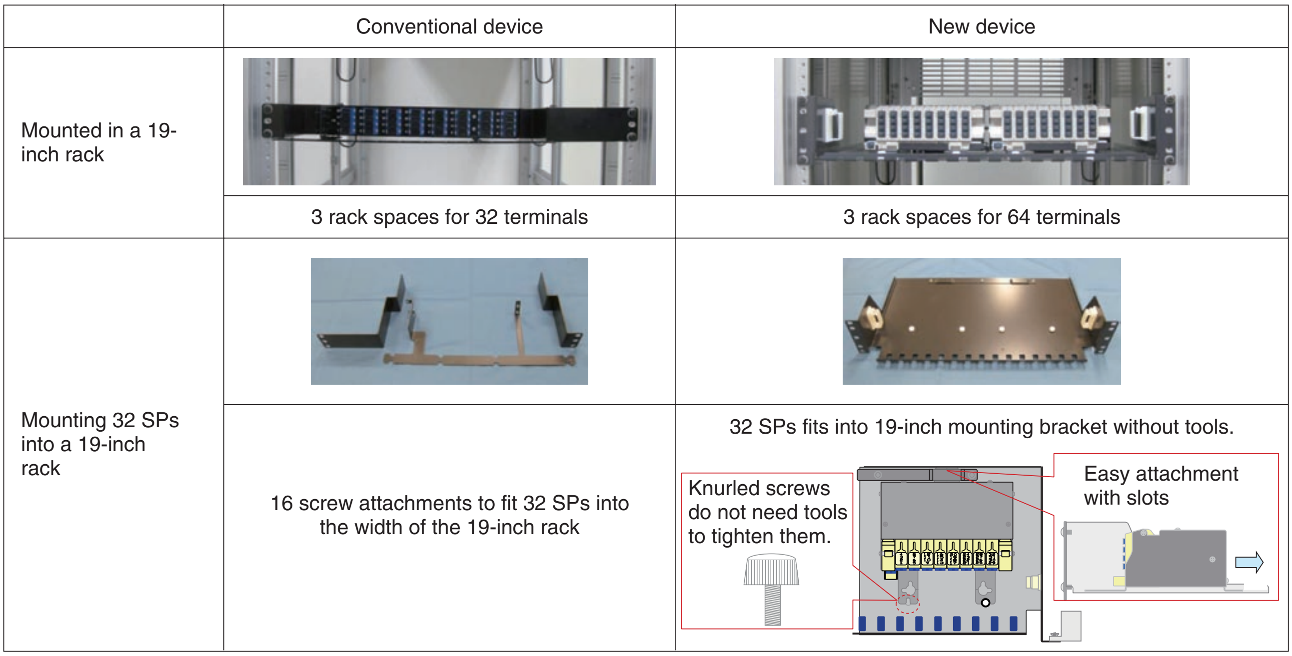

Fig. 5. Comparison of 19-inch rack mounting procedures.

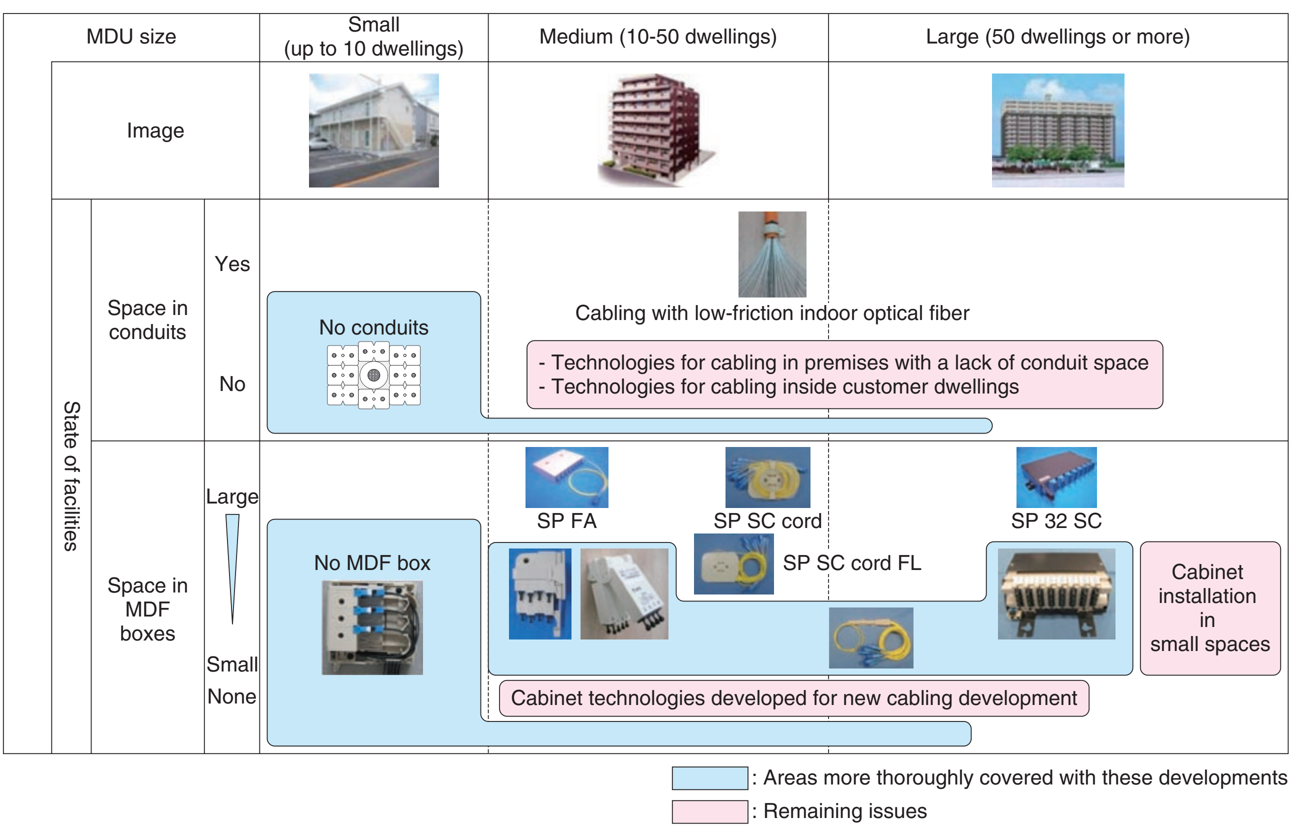

Fig. 6. Areas covered by the new developments and remaining issues.

conventional devices. We also designed a simple 19-inch rack installation method that does not require the use of specialized tools, unlike the usual hardware that requires many screw attachments to add an expansion SP.

An overview of the hardware is shown in **Fig. 4**. The new device that we have developed (SP 32 SC-S (slide/small)) fits in half the space required for the conventional device, so two of them, supporting 64 subscribers, can be installed in the 3U required to install and work on one conventional unit supporting 32 subscribers.

We also developed a 19-inch mounting bracket for the SP 32 19-inch rack installation (**Fig. 5**) to which the unit is secured by simply sliding it into position and securing with only two screws at the front. We think that it could reduce work time by up to 80%. We have also aimed for better workability by using knurled screws for tool-free tightening by hand and keyhole-type slot-in screw holes.

5. Further plans

These new cabling technologies can improve workability and expand the service provision coverage. However, although the uptake of optical fiber services such as FLET'S HIKARI continues at a phenomenal rate, it does not mean that 100% of services can be provided to all subscribers. As shown in **Fig. 6**, further work is needed to expand services to areas not covered by these new developments. With these new technologies as a base, we plan to continue working to develop indoor cabling technologies that can be applied to all MDUs as well as technologies that streamline the installation work required for optical fiber access services.

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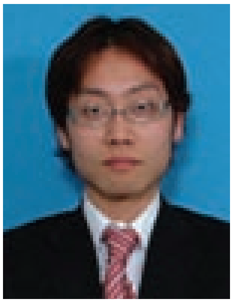

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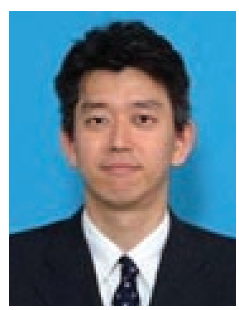

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Lower-cost Cicada-resistant Optical Fiber Drop Cable with Better Workability

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Abstract

We have developed a simpler type of cicada-resistant optical fiber drop cable that offers better workability and is less expensive than the older one. It resulted from a rethink of the protective walls and sheath design.

1. Introduction

Increases in the number of contracts for fiber-to-the-home (FTTH) services mean increases in the number of optical cabling installations, which are leading to the emergence of various issues with optical fiber drop cables. Since identifying damage due to cicadas in 2001, NTT has studied measures to prevent such damage. In 2007, cables with protective walls [1] were introduced to protect the optical fibers from this type of damage (**Fig. 1**). Although these cables are effective in preventing cicada damage, their more complicated structure means they are not as easy to work with as the conventional drop cables and they are more expensive. To solve these problems, we have developed a new low-cost optical fiber drop cable that is effective in preventing cicada damage and has good workability [2].

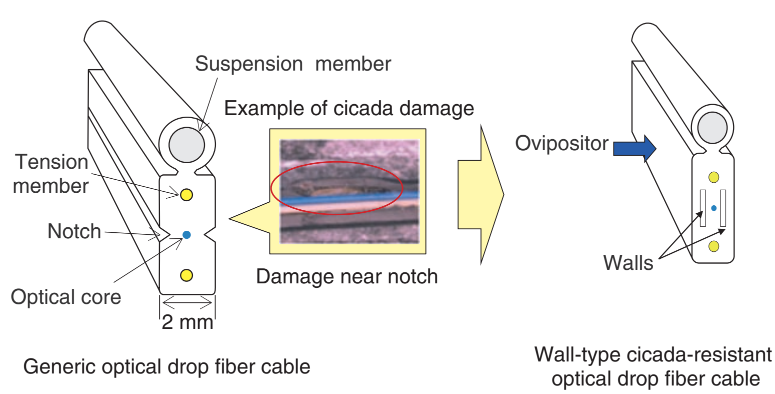

Fig. 1. Conventional cicada-resistant measures.

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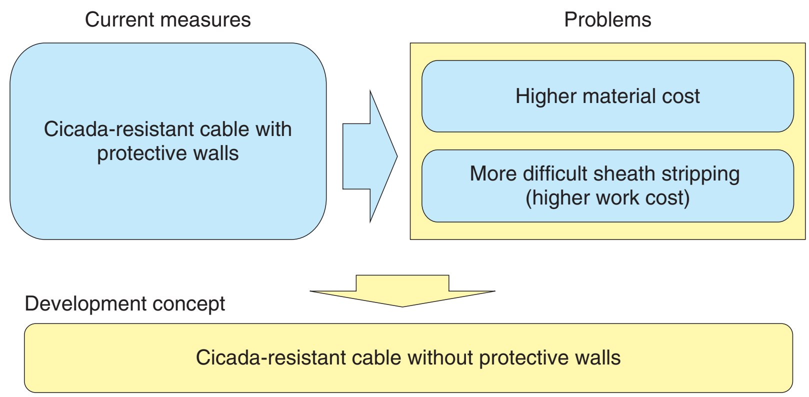


Fig. 2. Development concept.

2. Cicada damage and conventional anti-cicada measures

The cicada (*Cryptotympana facialis*) that causes the damage to drop cables is one of the larger species of cicada that inhabit Japan: it grows to a length of 6–7 cm. Preferring a warmer climate, the species can be found in western Japan. During spawning, the female normally bores a hole in a branch of deadwood by stabbing the wood with her sharply pointed ovipositor and then lays her eggs in the hole. Since drop cables are similar in size and are erected at a similar height to the branches that the cicadas prefer for egg laying, they can easily be mistaken for a branch by cicadas. The conventional drop cables had a sheath made from soft materials, which let the cicada gradually bore a hole into the cable and break the optical fiber. The cicadas tended to bore into the cable's notch, which was intended to make it easier to strip and expose the optical fiber core during installation work. To prevent cicadas from being attracted to the notch, the notch was eliminated from the cable, despite its usefulness. The hard protective walls on either side of the optical core meant that the cicadas were no longer able to bore deeply into the cable. The introduction of this cable has reduced the amount of damage from cicadas.

3. New development concept

Although it was effective, the cable with protective walls was more expensive to manufacture and harder to work with when it needed to be stripped to expose the core. The new development eliminates those protective walls while maintaining equally good cicada-resistance characteristics (**Fig. 2**). Instead of walls inside the cable, we have produced a simpler design based on a harder sheath material that is more difficult for cicadas to penetrate. This new cable is expected to have lower costs and better workability.

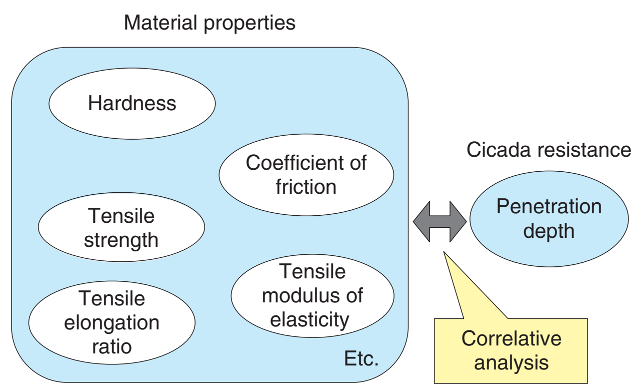


Fig. 3. Considering material properties.

4. Experiments

4.1 Method

Factors that give sheathing material its properties include hardness, tensile strength, tensile modulus of elasticity, tensile elongation ratio, and coefficient of friction. Since cicada resistance can be deduced from the ovipositor's penetration depth, we made several test cables with different properties to determine the right levels and combinations of these factors for cicada resistance. For each sample, we determined the relationship between the material properties and penetration depth (**Fig. 3**) [3].

To obtain data corresponding closely to actual conditions, we used live cicadas for our experiments. As shown in **Fig. 4**, we constructed a large cage into which we placed many cicadas captured outdoors. These were allowed to spawn. To ensure that the number of times cicadas laid eggs in each cable was accurately recorded, the cicadas in the cage were carefully observed and each spawning was counted. After the experiment, the cables were cut up and the penetration depths were measured radially from the surface. Examples of cicada penetrations are shown in **Fig. 5**.

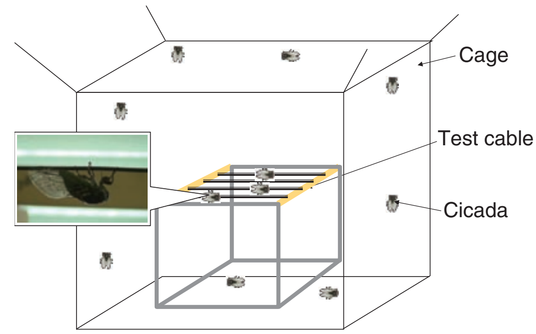
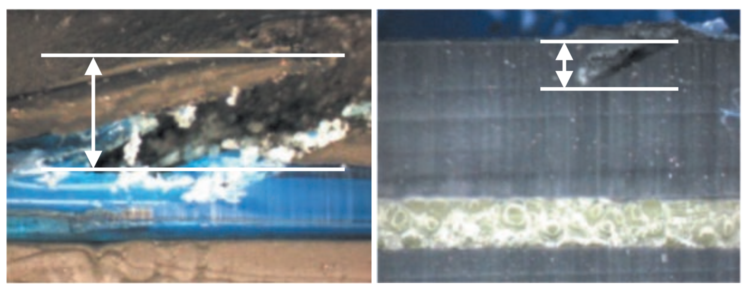


Fig. 4. Cicada cage experiment.



Reaches the fiber

Does not reach the fiber

Fig. 5. Penetration examples from the cage experiment.

4.2 Results

By testing various materials in the cicada cage experiment, we determined that hardness and tensile strength have an important effect on penetration depth. Hardness is important in determining the material's resistance to the concentrated external force applied to the cable by the cicada ovipositor. When considering a suitable hardness index for cicada spawning, we chose durometer hardness type D, which measures the repellent force of a needle-shaped indenter, an analog of the cicada ovipositor. As shown in **Fig. 6**, we found that the penetration depths became smaller as we increased the material's hardness.

Tensile strength is an index that can describe the material's toughness. We found that toughness, the material's ability to resist fracture, also plays an important role in preventing penetration by cicada ovipositors. A material's toughness is expressed as the amount of energy required to break it, which is effectively indicated by its tensile fracture strength.

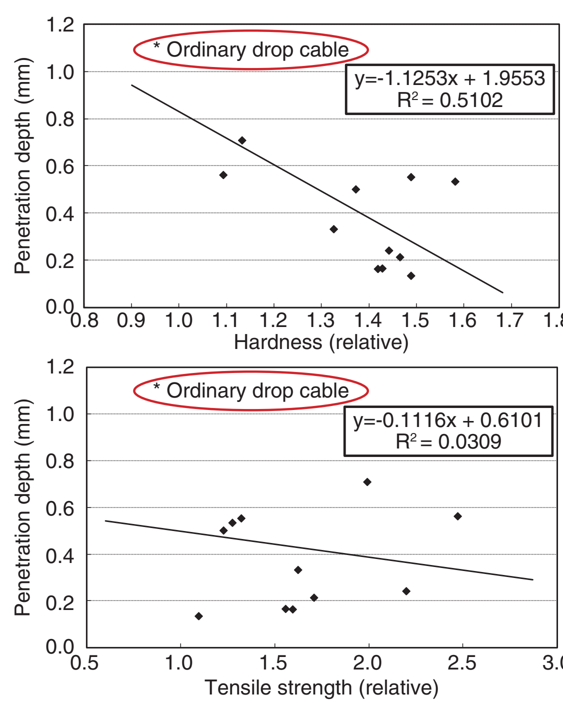


Fig. 6. Relationships between material properties and penetration depth.

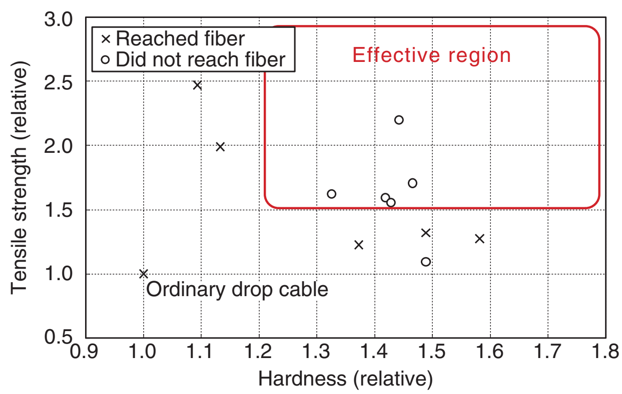


Fig. 7. Relationship between hardness and tensile strength for fiber-reaching and fiber-non-reaching penetration depths.

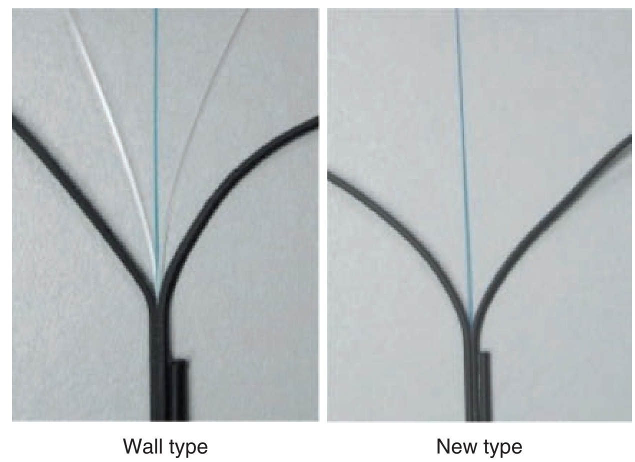


Fig. 9. Appearance of the new cable type.

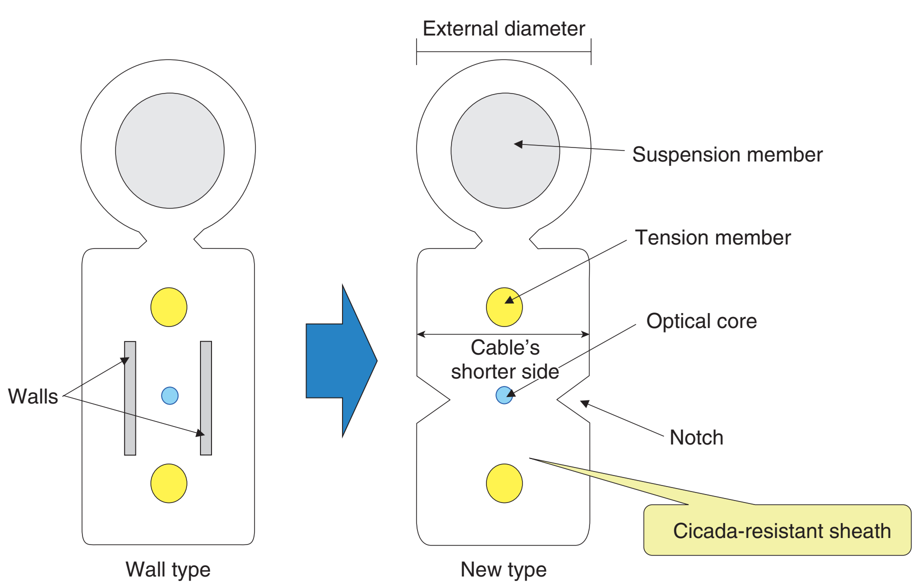


Fig. 8. Structure of the new cable type.

Although we were able to determine by experiment the importance of these two factors, we were unable to decisively determine effective ranges for them individually. However, when we analyzed the relationship between hardness and tensile strength, we were able to clarify an effective region (**Fig. 7**). We found that the combination of hardness to prevent penetration by the cicada ovipositor and toughness to withstand the repeated stabbing of the ovipositor was effective against cicada attack.

5. New cable

Through the process and experimentation described above, we were able to develop an optical fiber drop cable using a sheath material resistant to cicada attack [4]. The cable structure and appearance are shown in **Figs. 8** and **9**. The main characteristics of the new cable are described below.

5.1 Basic structure

As in the conventional drop cables, the suspension

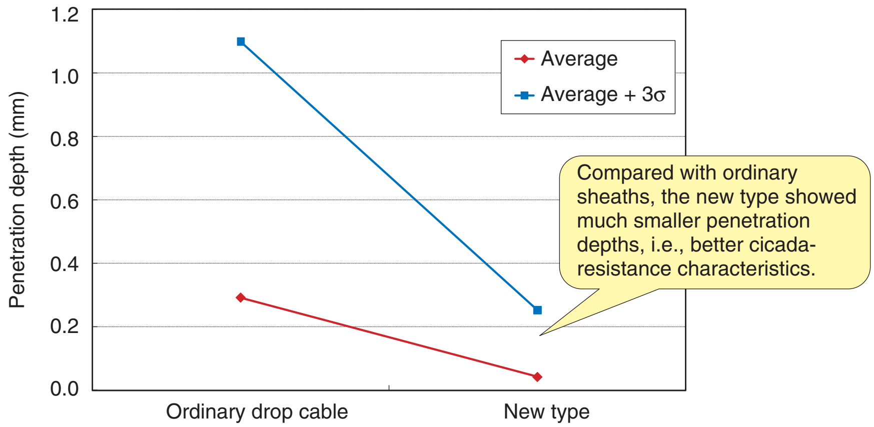


Fig. 10. Cicada-resistance characteristics of the new cable type.

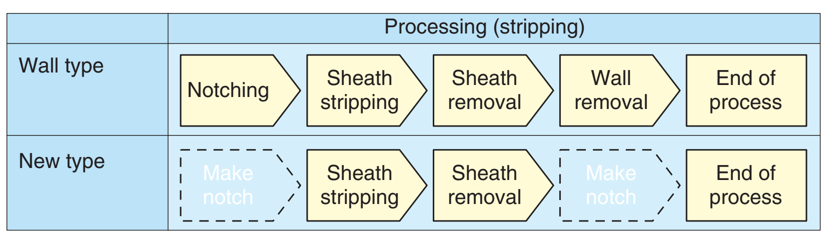


Fig. 11. Simplified processing.

member and cable are incorporated as a single unit. The suspension member is a 1.2-mm-diameter single steel wire, as in the conventional cable, and we kept its external diameter the same as the shorter side of the rectangular cable. To make the cable compatible with existing closures and sheath-holding connectors etc., we retained the dimensions of existing drop cables. We also included notches in the middle of the cable to enable easy sheath stripping.

5.2 Cicada resistance

Since maintaining cicada resistance was the most important consideration in this development, we verified the cable's reliability in terms of cicada resistance. We found that the ends of cicada penetrations into the cable were kept a satisfactory distance away from the optical fiber core. A comparison of penetration depths with a generic non-cicada-resistant cable is shown in **Fig. 10**. It clearly shows that penetrations in the new drop cable were much shallower.

5.3 Workability

In general, a harder sheath material means that sheath stripping is more difficult. However, to maintain good workability with the harder sheath material and ensure that the sheath is sufficiently cicada resistant, we optimized the notch shape and were able to create a cable that can be stripped in the normal manner.

6. Effects of this development

This new cicada-resistant cable has two main advantages. (1) Its simpler design eliminates the need for protective walls and thus lowers both material and manufacturing costs: it is more than 10% cheaper to produce. (2) Workability has been improved by including cable notches that enable easier sheath stripping to expose the optical fiber core; this means that technicians can use ordinary snips instead of specialized notching tools to strip the sheath. It also eliminates the wall-removal step (**Fig. 11**). This development produced a cable with the same cicada-resistance characteristics as the conventional cable type, but with a lower cost and better workability. It was first introduced in the fourth quarter of fiscal 2009 by NTT EAST and NTT WEST.

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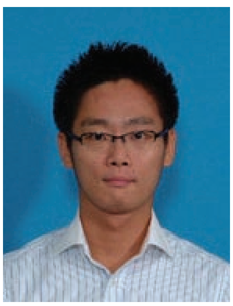

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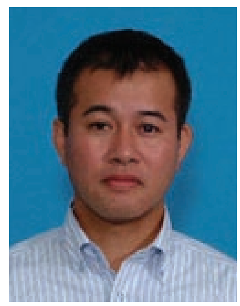

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Standardization Activities for Cloud Computing

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Abstract

This article describes standardization activities for cloud computing. Since cloud computing involves a wide range of technical and business elements, the targets of cloud computing standardization are diverse and many standards organizations are studying cloud computing focusing on their respective areas of expertise. The study areas can be broadly classified into: (i) framework development, terminology definition, use cases, and requirements identification, (ii) cloud configuration management, and (iii) inter-cloud federation. Cloud computing standardization was started by industrial organizations, which develop what are called forum standards. Since late 2009, de jure standards bodies, such as ITU-T and ISO/IEC JTC1, and ICT-oriented standards bodies (ICT: information and communications technology), such as IEEE (Institute of Electrical and Electronic Engineers) and IETF (Internet Engineering Task Force), have also begun to study cloud computing standardization. In the USA and Europe, government-affiliated organizations are also discussing it.

1. Introduction

Cloud computing represents a form of computing in which a user accesses networked-based information and communications technology (ICT) resources, such as servers, storage units and network devices, and services provided using those resources (cloud services) via the network without worrying about where the resources are actually located. It has been rapidly taken up by the business community over the last few years. This article describes standardization activities for cloud computing.

2. Importance of standardizing cloud computing

One advantage of using a cloud service is that users can use the service without owning their own ICT resources. Instead, they pay only for the amount of the service that they have actually used. The number of users of cloud-based consumer services, such as Amazon and Google, has been increasing rapidly in the last few years. Cloud computing is steadily making its way into enterprises' information systems and government systems.

Technological evolution of cloud computing has been so fast that cloud service providers have tried to differentiate their services by using their own service specifications. However, as cloud computing develops into widely used social infrastructures, serious problems associated with users being locked into a specific provider have arisen. Specifically, once an enterprise begins to use a specific provider's service, it finds it difficult to switch to another provider that offers more attractive service conditions and it faces the risk of its business being interrupted if the provider goes out of business.

To enable users to continue to use cloud services with confidence, activities for developing standards that ensure interoperability and portability between cloud services have been organized since around 2009.

3. Targets of cloud computing standardization

Cloud computing is a conglomerate of a wide variety of technologies, from distributed processing to virtualization. The types of service offered are also

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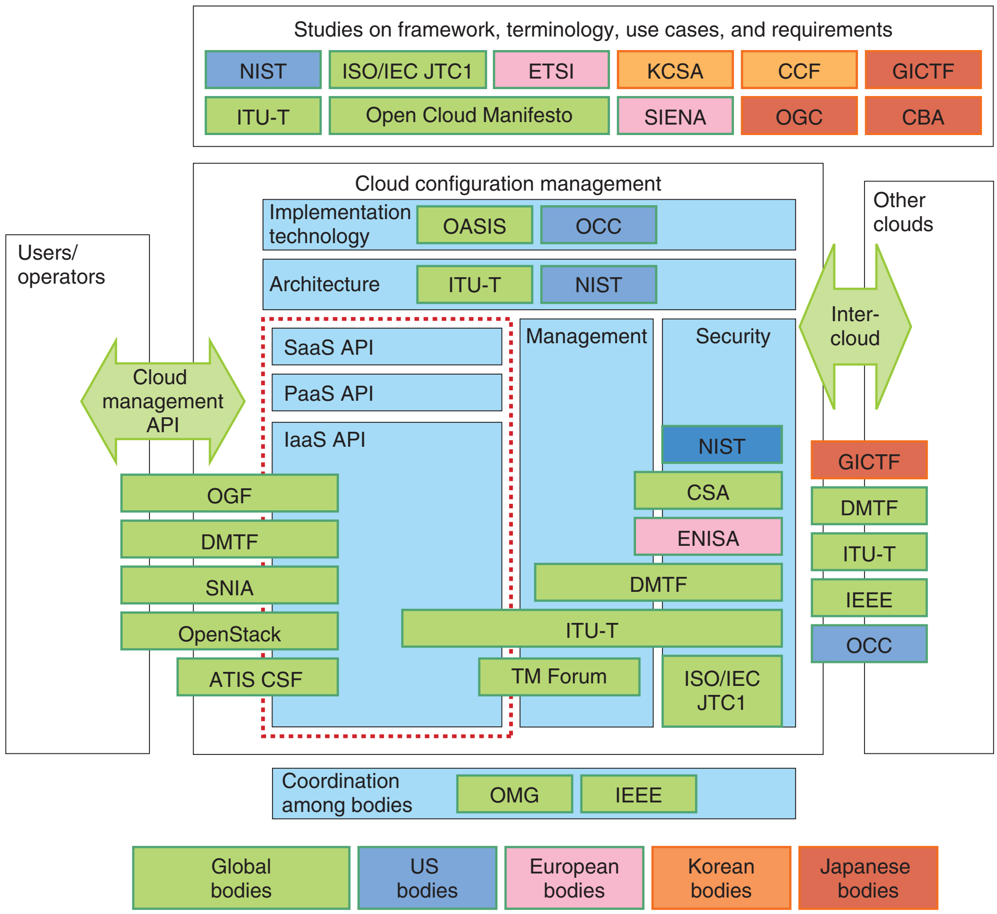

OGC: Open Government Cloud consortium
CBA: Cloud Business Alliance
SIENA: Standards and Interoperability for e-infrastructure Implementation initiative
KCSA: Korea Cloud Service Association
ATIS CSF: Alliance for Telecommunications Industry Solutions Cloud Services Forum

Fig. 1. Study areas for cloud computing standardization and major cloud computing standards bodies.

diverse: from infrastructure as a service (IaaS), in which ICT resources are provided as a service, and platform as a service (PaaS), in which database or other middleware programs are provided as a service, to software as a service (SaaS), in which application software, such as accounting programs and office suits, are provided as a service.

The targets of cloud computing standardization are so diverse that many standards organizations are studying cloud computing focusing on their respective areas of expertise. The different study areas and major cloud computing standards bodies related to each of them are shown in **Fig. 1**. Each body addresses issues that are so diverse that it is difficult to define precisely what issues it covers. However, the study areas for cloud computing can be broadly classified into the following three:

(i) Framework development, terminology definition, use cases, and requirements identification
(ii) Cloud configuration management
(iii) Inter-cloud federation.

3.1 Studies on framework development, terminology definition, use cases, and requirements identification

Since cloud computing involves a wide range of technical and business elements, it is crucial in advancing standardization to sort out these elements and define standard technical terms. A significant contribution in these areas is being made by the

National Institute of Standards and Technology (NIST), which first defined the now commonly used terms IaaS and SaaS.

To identify what needs to be standardized for cloud computing, many organizations take an approach of first discussing use cases and then sorting out requirements that need to be standardized. In addition to de jure standards bodies as the International Telecommunication Union, Telecommunication Standardization Sector (ITU-T) and International Organization for Standardization/International Electrotechnical Commission Joint Technical Committee 1 (ISO/IEC JTC1), major regional and national standards bodies in the USA, Europe, Korea, Japan, etc. are working on these standardization issues.

3.2 Studies on cloud configuration management

In the area of configuration management of cloud systems, many organizations are aiming to develop specifications for a standard application programming interface (API), an interface through which users and operators manage cloud systems.

The Open Cloud Computing Interface Working Group (OCCI-WG) of the Open Grid Forum (OGF) has defined and released an API for managing IaaS. The Open Cloud Standards Incubator (OCSI) of the Distributed Management Task Force (DMTF) has also specified an IaaS management API.

The Storage Networking Industry Association (SNIA) has formulated the specifications of the Cloud Data Management Interface (CDMI), which is an API for controlling storage units.

A non-standards organization is also active in this area. An open source community called OpenStack, which was established mainly by Rackspace and NASA in July 2010, makes the source code of its IaaS management software open to the public.

All the cloud configuration management APIs now under study are intended for IaaS. Few APIs are being studied for PaaS or SaaS. This can be attributed to the general recognition that PaaS and SaaS are in the competitive arena of business.

In addition, many organizations are studying reference architecture and security issues for cloud systems.

The Cloud Security Alliance (CSA) is defining the best practice in thirteen areas, ranging from security risks for cloud users and providers to governance and risk management that need to be defined in introducing cloud computing, electronic disclosure of legal evidence, compliance, and auditing.

3.3 Studies on inter-cloud federation

Studies on inter-cloud federation were started in the second half of 2009 so that multiple clouds can mutually give and take cloud resources to provide cloud services stably even when individual cloud systems become unable to provide service due to damage by a major natural disaster or a serious failure, or when their capacity is surpassed by the demand, or so that users can access two or more clouds seamlessly through ID-based federation (ID: identification).

The Global Inter-Cloud Technology Forum (GICTF), which was established in Japan in July 2009, is one of the first bodies to address the issue of inter-cloud federation. In June 2010, it issued a white paper that documented use cases and functional requirements for inter-cloud federation [1].

4. Current activities of major cloud-related standards bodies

Cloud computing standardization was started by industrial organizations, which develop what are called forum standards. Since late 2009, de jure standards bodies, such as ITU-T and ISO/IEC JTC1, and ICT-oriented standards bodies, such as IEEE (Institute of Electrical and Electronic Engineers) and IETF (Internet Engineering Task Force), have also begun to study it. In the USA and Europe, government-affiliated organizations are also discussing it. The activities of major forum standards bodies, ICT-oriented standards bodies, de jure standards bodies, and government-affiliated bodies are described below.

4.1 Forum standards bodies

There are two groups of forum standards bodies related to cloud computing. Those in the first group, including DMTF, OGF, and SNIA, have been active in the field of grids and distributed processing management and have newly added cloud computing to their agendas. The bodies in the second group, including OCC, CSA and GICTF, were newly founded to address cloud computing.

(1) DMTF (Distributed Management Task Force)

DMTF has defined the Open Virtualization Format (OVF), which is a standard virtual machine image format. It established OCSI in April 2009 and is studying standards that will allow interoperability between cloud systems. The Cloud Management Working Group (CMWG), in which VMware, Fujitsu, Oracle, and others are proposing a relevant API, was established in June 2010. DMTF issued a white paper on interoperability between cloud systems in November 2009 and another on use cases of cloud management and interactions in June 2010. Its Board Members include VMware, Microsoft, IBM, Citrix, Cisco, and Hitachi.

(2) OGF (Open Grid Forum)

OGF formed the OCCI-WG (WG: Working Group) in April 2009 and defined and released an API specification, OCCI [2], which makes possible lifecycle management of virtual machines and workloads through IaaS. OCCI is implemented in Europe's OpenNebula Project, etc. The main participants in OCCI are Fujitsu, EMC, and Oracle.

(3) SNIA (Storage Networking Industry Association)

SNIA established the Cloud Storage Technical Working Group in April 2009 and released CDMI, which is an interface specification for cloud storage data management. In October 2009, it formed a sub-working group called the Cloud Storage Initiative (CSI) to educate users and promote the cloud storage market through the Cloud BUR SIG (Cloud Backup and Recovery Special Interest Group) project. SNIA's membership includes EMC, IBM, Fujitsu, and Hitachi. Its Japan Chapter was established in 2010.

(4) OMG (Object Management Group)

OMG held a Cloud Standards Summit in July 2009 and inaugurated Cloud Standards Coordination, which is a round-table conference of cloud-related standards bodies. Participants in the Coordination currently include DMTF, OGF, SNIA, TM Forum (TeleManagement Forum), OASIS, OCC, CSA, ETSI, and NIST in addition to OMG.

(5) OASIS (Organization for the Advancement of Structured Information Standards)

OASIS established the Identity in the Cloud Technical Committee (IDCloud TC) in May 2010, surveyed existing ID management standards, and developed use cases of cloud ID management and guidelines on reducing vulnerability. It also developed basic security standards, such as SAML (Security Assertion Markup Language) and maintains liaison with CSA and ITU-T. The main members are IBM, Microsoft, and others.

(6) OCC (Open Cloud Consortium)

OCC is a nonprofit organization formed in January 2009 under the leadership of the University of Illinois at Chicago. It aims to develop benchmarks using a cloud testbed and achieve interoperability between cloud systems. Its Working Groups include Open-CloudTestbed, Project Matsu, which is a collaboration with NASA, and Open Science Data Cloud, which covers the scientific field. The main members include NASA, Yahoo, Cisco, and Citrix.

(7) Open Cloud Manifesto

Open Cloud Manifesto is a nonprofit organization established in March 2009 to promote the development of cloud environments that incorporate the user's perspective under the principle of open cloud computing. It published cloud use cases and requirements for standards as a white paper in August 2009. The latest version of the white paper is version 4.0 (V4) [3], which included for the first time the viewpoint of the service level agreement (SLA). The participants include IBM, VMware, Rackspace, AT&T, and TM Forum. A Japanese translation of the white paper is available [4].

(8) CSA (Cloud Security Alliance)

CSA is a nonprofit organization established in March 2009 to study best practices in ensuring cloud security and promote their use. It released guidelines on cloud security in April 2009. The current version is version 2.1 [5], which proposes best practices in thirteen fields, such as governance and compliance. The main members are PGP, ISACA, ENISA, IPA, IBM, and Microsoft. A distinctive feature of the membership is that it includes front runners in cloud computing, such as Google and Salesforce. A Japan Chapter of CSA (NCSA) was inaugurated in June 2010.

(9) CCF (Cloud Computing Forum)

CCF is a Korean organization established in December 2009 to develop cloud standards and promote their application to public organizations. Its membership consists of 32 corporate members and more than 60 experts. CCF comprises six Working Groups, including Media Cloud, Storage Cloud, and Mobile Cloud.

(10) GICTF (Global Inter-Cloud Technology Forum)

GICTF is a Japanese organization studying inter-cloud standard interfaces, etc. in order to enhance the reliability of clouds. As of March 2011, it has a membership of 74 corporate members and four organizations from industry, government, and academia. In

June 2010, it released a white paper on use cases of inter-cloud federation and functional requirements. The main members include NTT, KDDI, NEC, Hitachi, Toshiba Solutions, IBM, and Oracle.

4.2 ICT-oriented standards bodies

Major standards bodies in the ICT field have also, one after another, established study groups on cloud computing. These study groups are holding lively discussions.

(1) IETF

IETF had been informally discussing cloud computing in a bar BOF (discussions over drinks in a bar; BOF: birds of a feather) before November 2010 when, at IETF79, it agreed to establish the Cloud OPS WG (WG on cloud computing and maintenance), which is discussing cloud resource management and monitoring, and Cloud-APS BOF (BOF on cloud computing applications), which is mainly discussing matters related to applications. Since around the end of 2010, it has been receiving drafts for surveys of the cloud industries and standards bodies, reference frameworks, logging, etc.

(2) IEEE

IEEE formed the Cloud Computing Standards Study Group (CCSSG) in March 2010. It announced the launch of two new standards development projects in April 2011: P2301, Guide for Cloud Portability and Interoperability Profiles (CPIP) and P2302, Standard for Intercloud Interoperability and Federation (SIIF).

(3) TM Forum

In December 2009, TM Forum established the Enterprise Cloud Buyers Council (ECBC) to resolve issues (on standardization, security, performance, etc.) faced by enterprises when they host private clouds and thereby to promote the use of cloud computing. In May 2010, it started the Cloud Services Initiative, which aims to encourage cloud service market growth. The main members of this initiative are Microsoft, IBM, and AT&T.

4.3 De jure standards bodies

Since late 2009, major de jure standards bodies have taken up cloud computing as part of their study subjects. All these bodies are conducing a gap analysis based on the studies made by forum standards bodies in order to identify the target areas where standardization by de jure standards bodies is desired. Specific activities to develop recommendations are expected to start in 2011.

(1) ITU-T

In February 2010, ITU-T launched the Focus Group on Cloud Computing, which is discussing the benefits of clouds and target issues requiring standardization from the telecommunication perspective. The Group is currently developing six documents on topics such as the cloud ecosystem, functional architecture, cloud security, and utilization of networks in clouds. Afterwards, relevant Study Groups will develop recommendations for these issues.

(2) ISO/IEC JTC1

In its Sub Committee 38 (SC38) meeting held in November 2009, ISO/IEC JTC1 established a Study Group to study cloud computing. Its secretariat is provided by the American National Standards Institute (ANSI). The Study Group is classifying cloud computing, sorting out terminology, and maintaining liaison with other organizations. In Japan, SC38 Technical Committee was launched in February 2010. In addition, SC27 is studying requirements for Information Security Management Systems (ISMSs).

(3) ETSI (European Telecommunications Standards Institute)

ETSI has established a Technical Committee on grids and clouds. The TC Cloud has released a Technical Report (TR) on standards required in providing cloud services.

4.4 Government-affiliated bodies

Government-affiliated bodies in the USA and Europe are active in cloud-related standardization. Government systems constitute a large potential cloud market. It is highly likely that the specifications used by governmental organizations for procurement will be adopted as de facto standards.

(1) NIST

NIST is a technical department belonging to the U.S. Department of Commerce. "The NIST Definition of Cloud Computing", which was published in October 2009, is referred to on various occasions. NIST undertakes cloud standardization with five WGs. One of them, Standards Acceleration to Jumpstart Adoption of Cloud Computing (SAJACC), is intended to promote the development of cloud standards based on actual examples and use cases. It discloses a number of different specifications and actual implementation examples on its portal. It also discloses test results for the developed standard specifications.

(2) ENISA (European Network and Information Security Agency)

In November 2009, ENISA, an EU agency, released two documents: "Cloud Computing: Benefits, Risks and Recommendations for Information Security", which deals with cloud security, risk, and assessment and "Cloud Computing Information Assurance Framework", which is a framework for ensuring security in cloud computing.

5. Future of cloud standardization and NTT's activities

While many organizations are discussing cloud standardization, activities for consolidating their discussions are currently inadequate. Such activities are being undertaken only partially by Cloud Standards Coordination. The key issue will be how well the de jure standards bodies that have begun full-scale studies of cloud computing can collaborate with forum standards bodies.

Major front runners in cloud services, such as Amazon, Google, and Salesforce, are not participating in the cloud standards organizations mentioned in this article. There are not a few people who argue that it is too early for cloud standardization because it may impede technical innovation. It is unclear to what extent the standards developed by the standards bodies will be adopted by the market. This trend should be monitored carefully.

NTT believes that it is particularly important to standardize the external interfaces of cloud systems if we are to achieve interoperability and portability between cloud services. Specifically, such interfaces include interfaces for cloud users and those for cloud application developers and inter-cloud interfaces to allow federated operation between clouds.

NTT is currently participating in GICTF as a board member. NTT will make proposals to ITU-T and NIST on the basis of the work undertaken by GICTF on inter-cloud federation and other issues.

6. Conclusion

This article described standardization activities for cloud computing. NTT will promote the standardization of external interfaces of cloud systems, which will enable interoperability and portability between cloud services.

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Fault Cases and Countermeasures against Damage to Access System Facilities Caused by Wildlife

Abstract

Wildlife that damages access system facility is classified into three groups (rodents, insects, and birds). We have investigated the types of damage and have studied countermeasures. Such countermeasures will contribute greatly to reducing maintenance costs and improving the reliability of access system facilities. This article is the fifth in a bimonthly series on the theme of practical field information about telecommunication technologies. This month's contribution is from the Access Engineering Group, Technical Assistance and Support Center, Maintenance and Service Operations Department, Network Business Headquarters.

1. Introduction

Most of the access system facilities maintained by the NTT Group are exposed to the outside environment. This means that damage caused by rain, wind, lightning, and other natural phenomena can occur. In addition, more than a little damage is caused by wildlife. In this article, we report on the various kinds of damage due to wildlife and countermeasures to maintain the reliability of access system facilities in Japan.

2. Fault cases

Access system facilities between a central office and customer premises along with various kinds of wildlife that cause damage are schematically illustrated in **Fig. 1**. Wildlife that causes damage to access system facilities in Japan can be broadly classified into (1) rodents (squirrels, flying squirrels, rats, etc.), (2) insects (moth larvae, cicadas, ants, etc.), and (3) birds (crows, woodpeckers, etc.). These kinds of wildlife can be found throughout the country.

Most rodents are active all year (only a few hibernate), so the time when damage happens is not limited to a particular period. They mainly damage cables, especially small-diameter ones.

The degree of damage and the time that damage happens depends on the type and ecology of insects: moth larvae damage cables in early summer (June and July) before they become chrysalises, and cicada damage is caused specifically by adult females mainly from early summer to early fall (June to September).

Damage caused by birds mainly occurs during the breeding season. In particular, crows can cause extensive damage in the period from March to June.

2.1 Rodents

Rodents have incisors on both upper and lower jaws. These teeth continue to grow throughout their life, but rodents keep them to an appropriate length by gnawing on objects. Rodents can be broadly classified into the Sciuridae family that includes squirrels and flying squirrels and the Muridae family that includes black rats and house mice. They chew on telecommunication cables and connection closures and damage copper wires inside metallic cables and fibers inside optical cables. Damage caused by members of the Sciuridae family frequently occurs along aerial cable routes in mountainous areas and damage caused by members of the Muridae family often occurs in underground cable routes.

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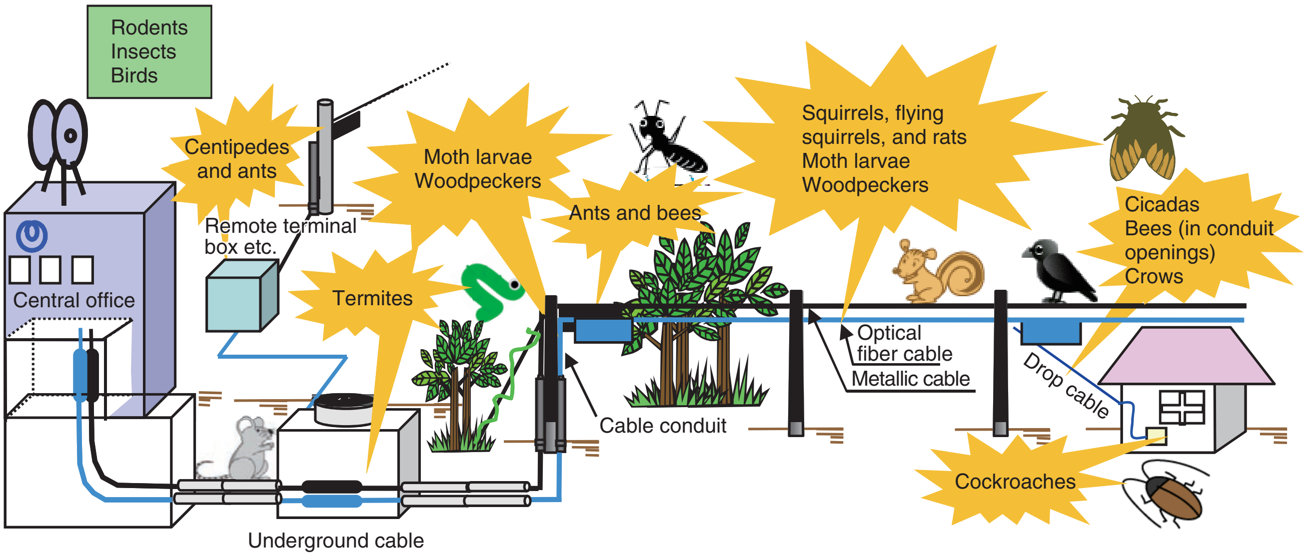

Fig. 1. Illustration of places in access system facilities that can be damaged by wildlife.

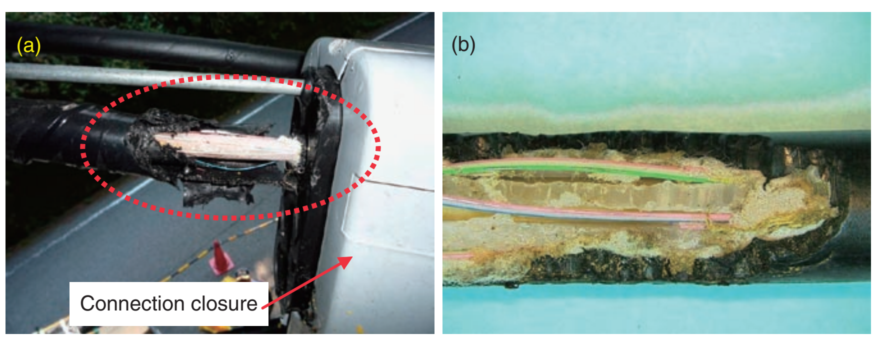

Fig. 2. Example of optical fiber cable damage (by a flying squirrel).

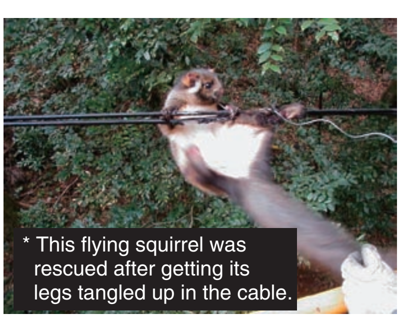

Fig. 3. Flying squirrel encountered in an area in which damage occurred.

Fig. 4. Damage to an FFA closure (by a squirrel).

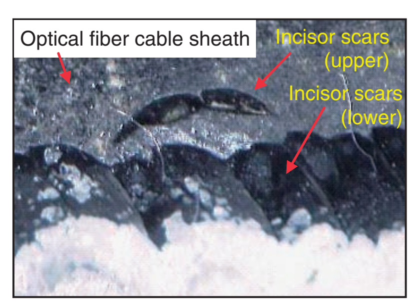

Fig. 5. Incisor scars left by squirrel (upper and lower).

An example of optical fiber cable damage (by a flying squirrel) is shown in **Fig. 2**, a flying squirrel encountered in an area in which damage occurred is shown in **Fig. 3**, damage to a fiber free access (FFA) closure (by a squirrel) is shown in **Fig. 4**, and a squirrel's incisor scars (upper and lower) are shown in **Fig. 5**.

2.2 Insects

2.2.1 Moth larvae

Moth larvae burrow into trees and the ground and live in holes. The larvae mistake telecommunication cables for plants. As a result, they can damage the cable by chewing. An example of damage to an optical fiber cable (by a moth larva) is shown in **Fig. 6**.

2.2.2 Cicadas

Cicadas generally lay their eggs in dead trees, avoiding living trees that might freeze over during the winter. They may also try laying their eggs in a drop cable by piercing the sheath with the ovipositor. This behavior can damage an optical fiber inside the cable. Cicada egg laying, cicada eggs, and damage to a drop cable (by a cicada) are shown in **Figs. 7–9**.

2.2.3 Other insects

Other insects like ants and bees may invade connection closures or cable conduits through very small gaps to build nests. This kind of activity can damage facilities, and there are examples of centipedes invading outdoor access system equipment and getting burned up by coming into contact with package terminals. Centipedes prefer tight spaces and can invade equipment through small gaps such as cable entryways. A centipede found inside an outdoor remote subscriber module is shown in **Fig. 10**.

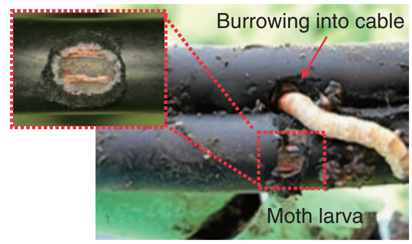

Fig. 6. Damage to an optical fiber cable (by a moth larva).

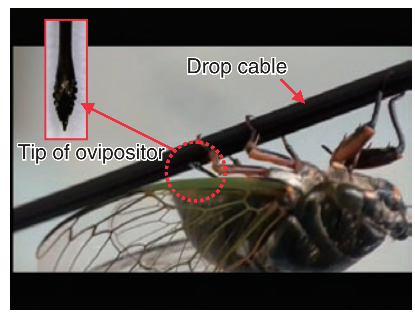

Fig. 7. Cicada egg laying.

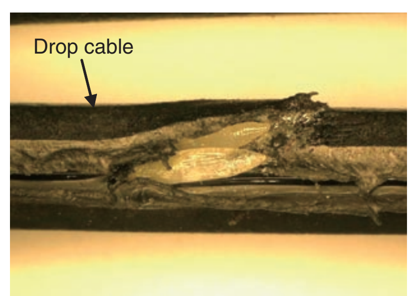

Fig. 8. Cicada eggs.

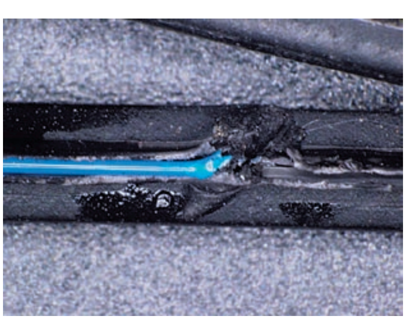

Fig. 9. Damage to a drop cable (by a cicada).

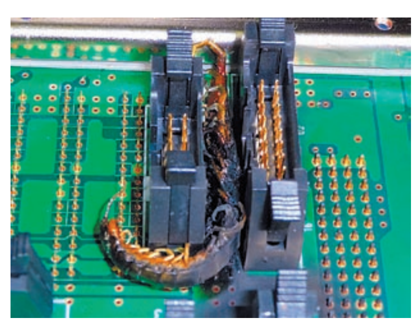

Fig. 10. Centipede (charred body) found inside an outdoor remote subscriber module.

2.3 Birds

2.3.1 Crows

In the breeding season before summer, a pair of crows will build a nest by gathering slim, sturdy materials like twigs or wires. During this time, these crows may use their beaks to bite around a drop cable's water-draining section (where the supporting wire separates from the optical fiber section) and damage the optical fiber inside the drop cable. An example of damage to a drop cable (by a crow) is shown in **Fig. 11**.

2.3.2 Woodpeckers

Woodpeckers engage in drumming (knocking on a tree trunk with their beaks) to prey upon insects or to assert territory or mating rights. They may also do this on telecommunication cables, opening up holes in the sheath and damaging copper wires. An example of damage to a metallic cable (by a woodpecker) is shown in **Fig. 12**.

3. Countermeasures

3.1 Rodents

One method for preventing the problem introduced above is to use high-strength-sheath cable that has an internal stainless-steel layer. Another effective method for preventing partial cable damage is to use a squirrel-proof cover or squirrel-proof tape (both have a layer of stainless steel affixed to a PVC (polyvinyl chloride) sheet). Connection closures can be protected by commercial protective covers. A high-strength-sheath cable, a squirrel-proof cover, squirrel-proof tape, an example of the construction of a squirrel-proof cover and squirrel-proof tape, and a protective cover for connection closures are shown in **Figs. 13–17**.

3.2 Insects

The countermeasures taken against moth larvae are basically similar to those for rodents. Another effective method for preventing cable damage is to eliminate intrusion paths. It is effective to use commercial anti-ivy products on the stay (**Fig. 18**), to remove ivy entangled on the stay and elsewhere, and to cut down trees in contact with cables to simply deny larvae a path to cables and other facilities. One countermeasure for protecting outdoor metallic cables against cicadas is to change to type-G cables equipped with a metal band. For optical fiber drop cables, NTT Laboratories has tried various ideas such as removing the V-notches on the surface of the cable's sheath and incorporating an internal barrier. Consequently,

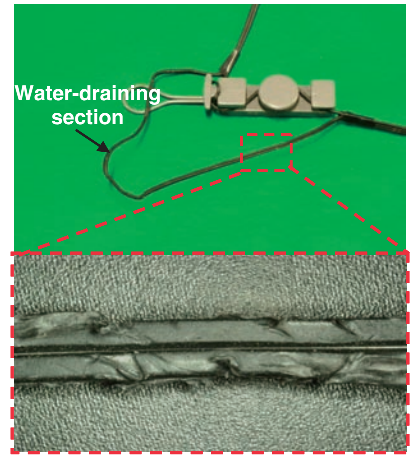

Fig. 11. Damage to a drop cable (by a crow).

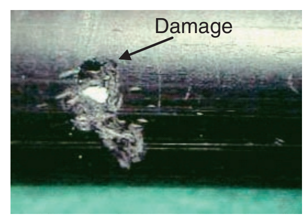

Fig. 12. Damage to a metallic cable (by a woodpecker).

Fig. 13. Appearance of high-strength sheath cable.

Fig. 14. Squirrel-proof cover.

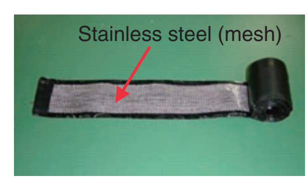

Fig. 15. Squirrel-proof tape.

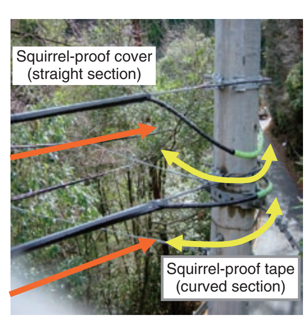

Fig. 16. Example of the construction of squirrel-proof cover and tape.

Fig. 17. Commercial protective cover for connection closures.

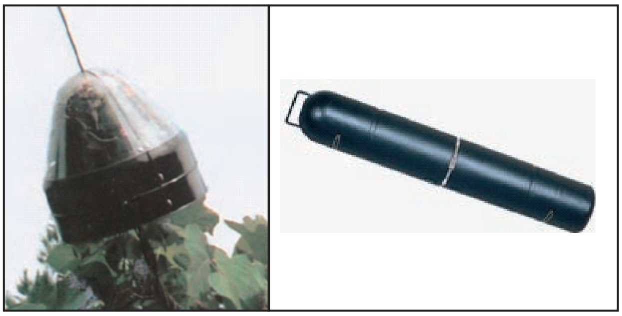

Fig. 18. Commercial anti-ivy products.

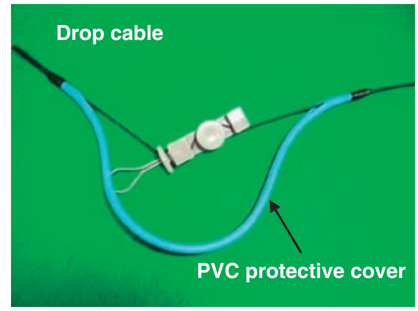

Fig. 19. PVC protective cover for optical cable.

Laboratories adopts optical fiber drop cable with a hardened sheath material that prevents the cicada's ovipositor from penetrating the cable [1].

Damage caused by other insects that invade facilities through small gaps can be prevented by simply blocking the intrusion path. In the outdoor equipment mentioned in this article, these gaps are likely to be found at cable entryways or ventilation holes. Cable entryways can be hermetically sealed and a metal mesh can be attached over ventilation holes.

3.3 Birds

An effective measure for preventing damage caused by crows is to wrap the most susceptible location—the water-draining section where the supporting wire separates from the optical fiber section—with a PVC protective cover, as shown in **Fig. 19**.

Damage caused by woodpeckers can be prevented by using a squirrel-proof cover and by installing squirrel-proof tape in the straight and curved sections of cable. It is also effective to use high-strength-sheath cable for aerial cables.

4. Conclusion

We reported on the various kinds of damage due to wildlife and countermeasures to maintain the reliability of the access system facilities in Japan. Wildlife may pose an ongoing risk to the operation of outdoor facilities. Therefore, it is important to apply efficient countermeasures. We would be pleased if the methods introduced here succeed in preventing wildlife damage.

Reference

[1] K. Nakano, K. Shiraishi, Y. Maehara, O. Inoue, A. Daido, K. Takamizawa, and T. Numata, "Lower-cost Cicada-resistant Optical Drop Cable with Better Workability," NTT Technical Review, Vol. 9, No. 6, 2011.
https://www.ntt-review.jp/archive/ntttechnical.php?contents=ntr201106ra2.html

External Awards

Best Interactive Award

Winners: Kyosuke Nishida[*1], Ryohei Banno[*2], Ko Fujimura[*1], and Takahide Hoshide[*1]

*1 NTT Cyber Solutions Laboratories

*2 Hokkaido University

Date: Mar. 1, 2011

Organization: The 3rd Forum on Data Engineering and Information Management

For "Tweet-Topic Classification on Twitter with Data Compression".

Published as: K. Nishida, R. Banno, K. Fujimura, and T. Hoshide, "Tweet-Topic Classification on Twitter with Data Compression," the 3rd Forum on Data Engineering and Information Management (DEIM 2011), A1-6, Izu, Japan.

Young Engineer's Award

Winner: Shin Kaneko, NTT Access Network Service Systems Laboratories

Date: Mar. 15, 2011

Organization: The Institute of Electronics, Information and Communication Engineers (IEICE)

For "Frequency-domain Optical CDM Employing Spectral M-ary Modulation Based on Electrical-domain Spatial Code Spreading"

This paper proposes a novel transmitter configuration that flexibly enhances the scalability for frequency-domain optical code-division multiplexing (CDM) based on electrical-domain spatial code spreading. The transmitter employs dummy data input and pre-biasing circuits. The dummy data input to the transmitter means that the total number of multiplexed binary data streams, comprising those that actually accommodate users/services and the dummy streams, remains constant. Pre-biasing circuits enable us to achieve high tolerance to multiple access interference by compensating for the nonlinearity of the M-ary modulation and improve the receiver sensitivity. Owing to the dummy data input, none of the parameters for prebiasing must be changed regardless of the number of users/services. Therefore, the proposed transmitter can flexibly enhance the scalability of optical CDM. The feasibility of the proposed transmitter is verified theoretically.

Published as: S. Kaneko, N. Miki, H. Kimura, and H. Hadama, "Frequency-domain Optical CDM Employing Spectral M-ary Modulation Based on Electrical-domain Spatial Code Spreading," Technical Report of IEICE, OCS (B-10-47), Vol. 110, No. 176, pp. 37–40, 2010 (in Japanese).

For "Spectral Efficiency Improvement in Frequency-domain Optical CDM Based on Electrical-domain Spatial Code Spreading"

Published as: S. Kaneko, N. Miki, H. Kimura, and H. Hadama, "Spectral Efficiency Improvement in Frequency-domain Optical CDM Based on Electrical-domain Spatial Code Spreading," Technical Report of IEICE (B-10-45) No. 2, p. 232, 2010 (in Japanese).

Papers Published in Technical Journals and Conference Proceedings

Heterostructure Growth of a Single-crystal Hexagonal AlN (0001) Layer on Cubic Diamond (111) Surface

K. Hirama, Y. Taniyasu, and M. Kasu

J. Appl. Phys. Vol. 108, No. 1, p. 013528, 2010.

We demonstrate heterostructure growth of a hexagonal AlN (0001) layer on cubic diamond (111) surface and investigate the interface structure in order to achieve AlN/diamond heterojunction devices. From the initial growth, the single-crystal AlN (0001) layer grows on the diamond (111) surface with an in-plane epitaxial relationship $[1010]_{\mathrm{AlN}}//[110]_{\mathrm{diamond}}$. A high-resolution transmission electron microscope image shows an abrupt interface. Misfit dislocations are distributed periodically at the heterointerface owing to the large lattice mismatch between AlN and diamond. Compared with the in-plane epitaxial relationship $[1120]_{\mathrm{AlN}}//[110]_{\mathrm{diamond}}$, $[1010]_{\mathrm{AlN}}//[110]_{\mathrm{diamond}}$ is energetically preferred because it has a higher bond density and, therefore, lower interfacial energy.

Extension of Secret Handshake Protocols with Multiple Groups in Monotone Condition

Y. Kawai, S. Tanno, T. Kondo, K. Yoneyama, K. Ohta, and N. Kunihiro

IEICE Trans. on Fundamentals of Electronics, Communications and Computer Sciences, Vol. 93, No. 6, pp. 1122–1131, 2010.

The Secret Handshake protocol allows members of the same group to authenticate each other secretly. That is, two members who belong to the same group can learn that the counterpart is in the same group, while non-members of the group cannot determine whether or not the counterpart is a member of the group. Yamashita and Tanaka proposed Secret Handshake Scheme with Multiple Groups (SHSMG). They extended a single group setting to a multi-group setting where two members output "accept" if both members' affiliations of the multiple groups are identical. In this paper, we first show a flaw in their SHSMG and construct a new secure SHSMG. Second, we introduce the new concept of the Secret Handshake scheme "monotone condition Secret Handshake with Multiple Groups (mc-SHSMG)" in order to extend the "accept" condition. In our new handshake protocol setting, members can authenticate each other in the monotone condition (not only in the case where both members' affiliations are identical but also in the case where they are not identical). The communication costs and computational costs of our proposed mc-SHSMG are lower than for the trivial construction of it.

Proxiable Designated Verifier Signature

M. Ushida, K. Ohta, Y. Kawai, and K. Yoneyama

Proc. of Security and Cryptography (SECRYPT) 2010, pp. 344–353, Athens, Greece.

Designated Verifier Signature (DVS) guarantees that only a verifier designated by a signer can verify the validity of a signature. In this paper, we propose a new variant of DVS: Proxiable Designated Verifier Signature (PDVS), where the verifier can make a third party (i.e., the proxy) substitute for some of the verification process. In the PDVS system, the verifier can reduce his computational cost by delegating some of the verification process without revealing the validity of the signature to the proxy. In all DVS systems, the validity of a signature means that a signature satisfies both of the following properties: (1) the signature is accepted by a decision algorithm and (2) the signature is confirmed to have been generated by the signer. In the PDVS system, the verifier can make the proxy substitute for only the checking of property (1). In the PDVS model, we divide the verifier's secret keys into two parts: one is a key for performing the decision algorithm, and the other is a key for generating a dummy signature, which prevents a third party from being convinced of property (2). We also define security requirements for PDVS and propose a PDVS scheme that satisfies all of the security requirements that we define.

Transmission Characteristics of Chirp-managed Direct Modulation over Hybrid Link of SMF and DSF

S. Usui, H. Iwashita, and T. Kubo

OptoeElectronics and Communications Conference (OECC) 2010, pp. 298–299, Sapporo, Japan.

A chirp-managed direct-modulated optical signal has a high tolerance for fiber dispersion. However, its tolerance for nonlinear effects is not clear. We thus investigated its applicability to a hybrid link of SMF and DSF without line amplifiers.

Diamond Semiconductor and Its Prospects for High RF Power Devices

M. Kasu

Material stage, Technical Information Institute Co. Ltd., Vol. 10, No. 7, pp. 58–61, 2010 (in Japanese).

The present status and prospects of diamond RF power devices are reviewed.

Cross-realm Password-based Server Aided Key Exchange

K. Yoneyama

WISA, KIISC, Lecture Notes in Computer Science, 2011, Vol. 6513, pp. 322–336, Jeju, Korea.

In this paper, we extend password-based server aided key exchange (PSAKE) to the cross-realm setting which enables two clients in two different realms with different passwords to exchange a session key through their corresponding servers, i.e., there are two servers. We cannot simply apply the previous security model of PSAKE to the cross-realm setting because there is a difference between the security properties that can be captured in the previous setting and in the new setting. Therefore, we define a new formal security model of cross-realm PSAKE. Our model captures all desirable security requirements, like resistance to leakage of ephemeral private keys, to key-compromise impersonation, and to undetectable on-line dictionary attack. Furthermore, we propose a concrete construction of cross-realm PSAKE with the optimal number of rounds for a client that is secure in the sense of our model. Our scheme assumes no pre-established secure channels between different realms unlike previous schemes, but just authenticated channels between different realms.

Circulator-free Reflection-type Tunable Optical Dispersion Compensator Using Cascaded Arrayed-waveguide Gratings

Y. Ikuma, T. Mizuno, H. Takahashi, and H. Tsuda

Optical Communication (ECOC) 2010, Vol. We. 8. E. 7, pp. 1–3, Torino, Italy.

A tunable optical dispersion compensator that uses cascaded arrayed-waveguide gratings and an integrated phase shifter is reported. It has a reflective configuration but does not require a circulator. The dispersion is successfully controlled from +142 to +1148 ps/nm.

Observation of n-type Conduction in Arsenic-doped CVD Diamond

M. Kasu

Proc. of Int. Symp. on Compound Semicond 2010, Vol. 1, No. 1, p. 1, Takamatsu, Japan, 2010.

We achieved n-type diamond by using arsenic doping.

Increase of Hole Concentration of H-terminated Diamond and Its Application to FET

K. Michal and M. Kasu

Proc. of Int. Symp. on Compound Semicond 2010, Vol. 1, No. 1, p. 1, Takamatsu, Japan, 2010.

We improved H-terminated diamond FET by NO_2 adsorption.

Indifferentiable Security Reconsidered: Role of Scheduling

K. Yoneyama

Proc. of ISC'2010, Lecture Notes in Computer Science, No. 6531, pp. 430–444, Florida, CA, USA.

In this paper, the substitutability of the indifferentiability framework with non-sequential scheduling is examined by reformulating the framework through applying the Task-PIOA framework, which provides non-sequential activation with oblivious task sequences. First, the indifferentiability framework with non-sequential scheduling is shown to be able to retain the substitutability. Next, this framework is shown to be closely related to the reducibility of systems. Finally, two modelings with sequential scheduling and non-sequential scheduling, respectively, are shown to be mutually independent. Thus, the importance of scheduling in the indifferentiability framework is clarified.

Hierarchical ID-based Authenticated Key Exchange Resilient to Ephemeral Key Leakage

A. Fujioka, K. Suzuki, and K. Yoneyama

Proc. of IWSEC 2010, Lecture Notes in Computer Science, No. 6434, pp. 164–180, Kobe, Japan.

In real applications of (public key-based) cryptosystems, hierarchical structures are often used to distribute the workload by delegating key generation. However, there have been few previous studies about such a hierarchical structure in the ID-based authenticated key exchange (AKE) scenario. In this paper, we introduce the first hierarchical ID-based AKE that is resilient to ephemeral secret key leakage. We provide a formal security model for hierarchical ID-based AKE. Our model is based on eCK security to guarantee resistance to leakage of ephemeral secret keys. We also propose an eCK-secure hierarchical ID-based AKE protocol based on a hierarchical ID-based encryption.

Universally Composable NBAC-based Fair Voucher Exchange for Mobile Environments

K. Yoneyama, M. Terada, S. Hongo, and K. Ohta

Proc. of IWSEC 2010, Lecture Notes in Computer Science, No. 6434, pp. 42–59, Kobe, Japan.

Fair exchange is an important tool to achieve “fairness” of electronic commerce. Several previous schemes satisfy universally composable security which provides the property of security preservation over complex networks like the Internet. In recent years, as the demand for electronic commerce has increased, fair exchange for electronic vouchers (e.g., electronic tickets and money) to obtain services or contents has been in the spotlight. The definition of fairness for electronic vouchers is different from that for general electronic items (e.g., duplicated use of exchanged electronic vouchers by one user should be prevented). However, although there are universally composable schemes for electronic items, there have been no previous studies for electronic vouchers. In this paper, we introduce a universally composable definition of fair voucher exchange that represents ideal functionality for fair voucher exchange. Also, we prove the equivalence between our universally composable definition and the conventional definition for electronic vouchers. Thus, our formulation of the ideal functionality is justified. Finally, we propose a new fair voucher exchange scheme from non-blocking atomic commitment as a black-box, which satisfies our security definition and is adequate for mobile environments. Because general building blocks are instantiated with known practical ones, our scheme can also be practical because it is implemented without a trusted third party in usual executions.

Enhancement and Stabilization of Hole Concentration of Hydrogen-terminated Diamond Surface Using Ozone Adsorbates

M. Kubovic and M. Kasu

Jpn. J. Appl. Phys. Vol. 49, No. 110208, 2010.

The p-type conductivity of H-terminated diamond surface can be linked to adsorption of a specific gas species on the surface. O_3, NO_2, NO, and SO_2 were identified as adsorbates that induce holes on H-terminated diamond surface. Among them, exposure to O_3 increases hole concentration the most. The increased concentration remains high even after exposure to the gas has stopped, indicating that ozone is the most stable adsorbent. X-ray photospectroscopy spectra of O_3-adsorbed H-terminated diamond surface show partial oxidation of the surface and upward band bending and are very similar to those of NO_2-exposed diamond surfaces.

Arsenic-doped n-type Diamond Grown by Microwave-assisted Plasma Chemical Vapor Deposition

M. Kasu and M. Kubovic

Jpn. J. Appl. Phys. Vol. 49, No. 110209, 2010.

We grew n-type arsenic (As)-doped single-crystal diamond layers using tertiarybutylarsine as an As source. The n-type conduction of the As-doped layers was confirmed both in Hall measurements and from the current–voltage characteristics of the diodes. In the As-doped layers, electron concentration increased with As concentration in the layers. The ionization energy of the As donor decreased from 1.6 to 0.7 eV as As concentration increased from 1×10^{17} to 9×10^{19} cm^{-3}. A diamond p–n junction diode with an n-type As-doped layer exhibited a rectification ratio of ~1000 at ±10 V at room temperature.

Structure of Rat Ultrasonic Vocalizations and Its [*sic*] Relevance to Behavior

N. Takahashi, M. Kashino, and N. Hironaka

PLoS ONE, Vol. 5, No. 11, pp. e14115–14122, 2010.

Rats are known to emit ultrasonic vocalizations (USVs). These USVs have been hypothesized to hold biological meaning, and the relationship between USVs and behavior has been extensively studied. However, most of these studies looked at specific conditions, such as fear-inducing situations and sexual encounters. In the present experiment, the USVs of pairs of rats in ordinary housing conditions were recorded and their features were examined. Three clusters of USVs in the 25-, 40-, and 60-kHz ranges were detected, which roughly corresponded to fighting, feeding, and moving, respectively. We analyzed sequential combinations of two or more clusters using a state transition model. The results revealed a more specific correspondence between the USVs and behaviors, suggesting that rat USV may work as a type of communication tool.

NTT Communication Science Laboratories at TRECVID 2010 Content-based Copy Detection

R. Mukai, T. Kurozumi, K. Hiramatsu, T. Kawanishi, H. Nagano, and K. Kashino

Proc of TRECVID 2010, NIST, Vol. 1, No. 1, pp. 340–349, Gaithersburg, MD, USA.

In this paper, we describe our approaches that were tested in the TRECVID 2010 Content-Based Copy Detection (CBCD) task. We introduce a method consisting of a feature degeneration and sparse feature selection process for video detection tasks, which is highly robust as regards video signal distortion. For audio detection tasks, we adopt a method based on spectral partitioning to cope with additive interfering sounds. Both methods are key techniques for our Robust Media Search (RMS) technology, which has already been deployed for various commercial services. Evaluation results show the effectiveness of our methods.

Diamond/nitride Semiconductor Heterostructure: Growth and Properties

K. Hirama, Y. Taniyasu, and M. Kasu

Journal of the Surface Science Society of Japan, Vol. 31, No. 12, pp. 657–666, 2010 (in Japanese).

Diamond/III-V nitride semiconductor heterostructure appears promising not only for high-efficiency deep-UV light emitting diodes (LEDs) but also for high output power field-effect transistors (FETs). However, diamond has a diamond crystal structure, while III-V nitride semiconductors have a wurtzite crystal structure. Due to the difference in the crystal structures, single-crystal III-V nitride growth on diamond substrate has been difficult. In this study, we obtained single-crystal aluminum nitride (AlN) (0001) layers on diamond substrates by using the (111) diamond surface orientation and preventing the formation of the interface layer. We revealed the heteroepitaxial growth mechanism and proposed a model of the atomic arrangement at the diamond/AlN heterointerface. Furthermore, we demonstrated a p-type diamond/n-type AlN heterojunction diode and successfully observed band-edge emission from diamond. In addition, an AlGaN/GaN heterostructure with a two-dimensional electron gas (2DEG) was grown on diamond (111) by using the single-crystal AlN buffer layer.

Fabrication of p-n Junction Diamond Diodes with n-type Arsenic-doped Diamond

M. Kasu and M. Kubovic

MRS Meeting, the Materials Research Society (MRS), Vol. 1, No. 1, p. 1, Boston, MA, USA, 2010.

We achieved n-type diamond doped with arsenic and fabricated p-n junction diodes.

Effect of NO_2 and Its Related Molecules on Increasing Hole Concentration in Hydrogen-terminated Diamond

M. Kasu and M. Kubovic

MRS Meeting, the Materials Research Society (MRS), Vol. 1, No. 1, p. 1, Boston, MA, USA, 2010.

We investigated the increase in hole concentration caused by molecular adsorption.

Strongly Secure Two-pass Attribute-based Authenticated Key Exchange

Kazuki Yoneyama

Proc. of Pairing, Lecture Notes in Computer Science, 2010, Vol. 6487/2010, pp. 147–166, Yamanaka-onsen, Japan.

In this paper, we present a two-party attribute-based authenticated key exchange scheme that is secure in a stronger security model than the previous models. Our strong security model is a natural extension of the eCK model, which is for PKI-based authenticated key exchange, into the attribute-based setting. We prove the security of our scheme under the gap Bilinear Diffie-Hellman assumption. Moreover, while the previous scheme needs a three-pass interaction between parties, our scheme needs only a two-pass interaction. In a practical sense, we can use any string as an attribute in our scheme because the setup algorithm of our scheme does not depend on the number of attribute candidates (i.e., the setup algorithm outputs constant-size parameters).
